TASCAM

TEAC Production Products

244

.PORTASTUDIO

OWNER'S MANUAL

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- dbx noise reduction system made under license from dbx, incorporated. The name "dbx" and the dbx symbol are trademarks of dbx, Incorporated.
- Change in specifications and features may be made without notice or obligation.

WARNING: TO PREVENT FIRE OR SHOCK HAZARD, DO NOT EXPOSE THIS APPLIANCE TO RAIN OR MOISTURE.

This apparatus has a serial number located on the rear panel. Please record the model number and serial number and retain them for your records.

Model number _____

Serial number _____

INTRODUCTION TO THE MANUAL, AND HOW TO USE IT

The Model 244 PORTASTUDIO combines all the major components of the modern multitrack recording studio into one compact and portable unit. Even though it is small, it represents an enormous amount of high technology.

Basic Track Recording, Overdubbing in "Sync" and Remixing of 4 channels to mono or stereo can be accomplished with a minimum of extra equipment. For example, a songwriter working alone with the PORTASTUDIO would need only a mic, a mic cable and a headset to produce a complex demonstration tape. Of course, the instruments to make the music, skill in playing and a song to record are also necessary, so our large amount of technology can only be of use to the artist as a part of the entire creative process. The 244 PORTASTUDIO and its technology can enter the picture only after much hard work has already taken place.

At TASCAM, we recognize the fact that multichannel recording is never completely a scientific process. It is always a combination of many disciplines, and the products that we design and build provide only the potential for good results, not a guarantee.

When you purchase the PORTASTUDIO, we can supply you with the Technology, but in order to exploit it in the service of your Art you must expect to spend some time in learning what knob to turn and what switch to set for each function you require. You can help yourself by reading this Owner's Reference Manual slowly and carefully.

This manual presumes nothing about your past experience with the multichannel process except a willingness to learn. An expert in the field might see the significance of a control or jack with just a simple statement of its location, but in this manual we will point out "the obvious" many times just to make sure that you see the point. There is much to discuss, and we have tried to make these instructions as SIMPLE AS THE TECHNOLOGY WILL ALLOW.

The Logic Used in the Manual

In preparing this Manual, we started by asking ourselves these questions;

What Does the First Time User Want to Do?

Begin recording as soon as possible. In order to do this, the basic connections must be made, some additional equipment must be plugged in correctly, and some decisions about function must be made. You must deal with the technology, or the unit will do nothing useful.

What Should We Tell the First Time User About the PORTASTUDIO When We Begin?

Even an expert engineer will need some instruction at this point, so we must make decisions about the format of our manual. Rather than having separate sections, one with a great deal of complex analysis and another with a simplistic approach, we have decided to combine the two styles in one continuous job-related fashion.

Why? Consider these things:

First, this manual may be the only "experience" that you can draw upon. We have said that we will explain the benefit of each control as well as the way to "turn" or set it, and the science behind its function. If you have little or no practical experience, you will not easily find the advice you need unless we place it in a sequence that does not depend on prior knowledge of the process.

Second, the terms that we use in the index may not relate the use of the PORTASTUDIO or to the field that you already know. For example, a musician will call the beginning of the tape the "top", because it relates to the top of a piece of music paper. An audio-visual producer would refer to the beginning of the program as "home" as in "return home" when thinking of the rewind function. This variation in the "jargon" makes the use of a conventional Index or Table of Contents difficult, and you have no guarantee of finding the help you need easily. Our job related sequence of instruction is self indexing to a much larger extent. It is laid our this way:

1. COMPLETE ANALYSIS: The manual begins at the MIC/LINE INPUT connector on the mixer section back panel and goes through each circuit in the order in which they are wired, BUT THERE IS ALSO.....
2. A SIMPLE ILLUSTRATED GUIDE: The complex text is augmented with a group of large illustrations that have sufficient caption material to allow you to work more rapidly. If you have no troubles in following the "short form" instructions in the first large illustration, you can flip the pages to the next large drawing and continue the process. If you require more explanation of a circuit, simply read the "fine print" on the pages that come before and after the drawing that shows the function you are having trouble with.

The major topic groups in the manual are "job related" to the multitrack recording process and appear in this order.

1. **WARNINGS AND CAUTIONS**
   Before you begin, it is wise to find out what the unit will not do. Although it is highly flexible, it is not without limitations. Technology is not magic, read this section first and save time.
2. **RECORDING THE FIRST TRACK**
   The large foldout illustration in the middle of this section shows how to set up the PORTASTUDIO to record a "one mic" first track without any fancy "extras".
3. **MAKING THE FIRST OVERDUB**
   The large foldout illustration in this section shows the setup you will need in order to make the PORTASTUDIO "Overdub" ready.
   A. How to set up the simultaneous record and play aspect of the recorder section.
   B. Which on the mixer to use to "mix" the old (already recorded) and new (not yet recorded) signals together for a monophonic headphone "Cue" so you can hear what you are doing. This is also a basic layout, with no fancy "tricks".
   C. Making the second and third overdubs will use the "B" section information with very few changes. Since this is the "mode" that you will spend the most time in, the "fine print" in this section contains detailed explanations of many special overdub related setups.
4. **FINAL MIX WHEN ALL RECORDING IS COMPLETE.**
   This large foldout illustration is somewhat different in its concept than the first two. To perform a "Remix" of 4 tracks of tape, you must have a completed master, and this implies a reasonable amount of experience with the system. Remix as a concept is the part of the multitrack process where the "tricks" are not really extras, so this foldout shows the connection of accessories such as echo and EQ devices, power amps and loudspeakers as well as the basic Remix control settings on the PORTASTUDIO itself.

These four headings cover the sequence of the multitrack process in the broad sense, and can be said to apply to almost any use of an "overdub" capable system. In the "fine print" of each section you will find the specific examples of what each function is supposed to do for the "art" as well as how the circuit "works" in the technical sense. These "art" oriented examples of use will help make the fine points of the multitrack process more understandable to the first time user.

The First Time User

It is likely that more than one person will be involved in the recording process sooner or later, but we have written this owner's manual with the assumption that the first time user will start "all alone". Solo operation of this unit has been used in all the operating examples given. If you can comprehend the "solo" use of the PORTASTUDIO, you should have no troubles converting the logic of the unit to accommodate the extra "studio staff".

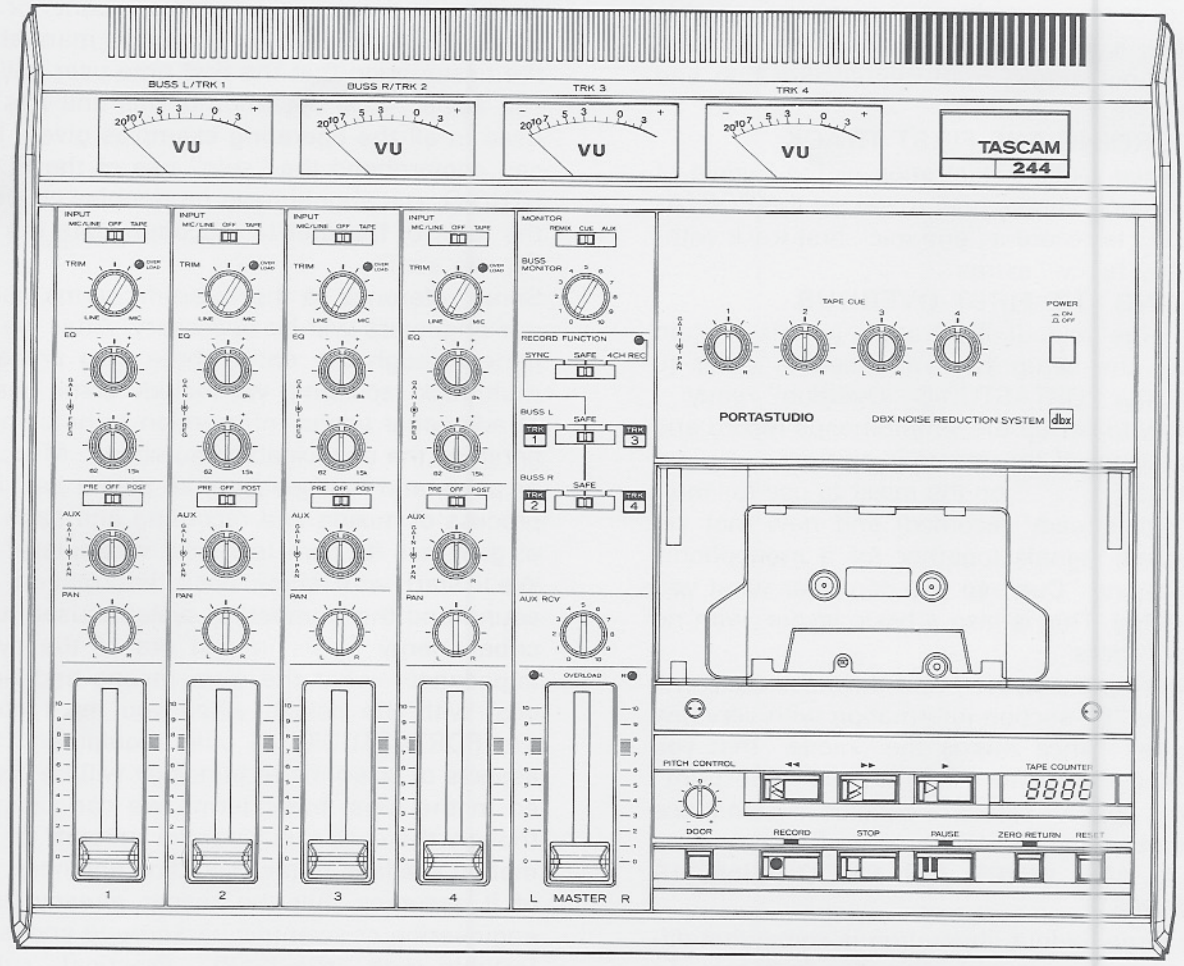
Some reference to the scientific terms used by our engineers will be necessary. After we have gone through the entire three step process of multitrack recording, we include some chapters of additional useful information to help you understand the unavoidable "numbers". Mics, power amps and loudspeakers all play a part in the process of mixing and recording and each piece of gear has its own technical vocabulary.

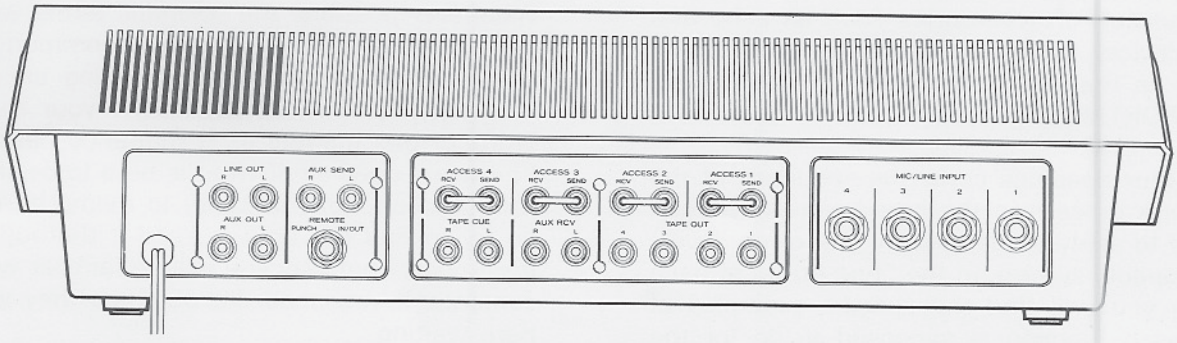
We'll start with some basic information about sound and the numbering systems used to describe energy levels in and out of the system, impedance—what the term means and how to deal with the details when you must connect the PORTASTUDIO to other equipment. Many aspects of scientific terminology will be discussed in the most basic terms we can use. Even though there is a substantial amount of information availabe to the recording engineer, much of it assumes that the reader already has an engineering or scientific background and is comfortable with "the math". Practical "rules of thumb" for the artist are not generally available, and in fact, to operate the PORTASTUDIO no degree in science is necessary. You don't have to build a mixer "from scratch" you just need to know how to find the right control function to get the job done.

Whenever possible, the scientific terms we use will be related to understandable common references. Understanding what is going on inside your equipment will help improve your sound.

Think of this manual as a reference handbook. You won't need all of what is here to begin, and it is certainly not necessary to memorize it, but do try to find the time to read it thoroughly at least once. That way you will be familiar with its contents. If you need the numbers they will be here waiting.

Good luck with your projects.





*NOTE: See page 52 for voltage conversion, and note for U.K. customers.

No matter what happens to the signal while it is being processed, it will eventually be heard once again by a human ear. So the process of converting a sound to an electrical quantity and back to sound again must follow the logic of human hearing.

The first group of scientists and engineers to deal with the problems of understanding how the ear works were telephone company researchers, and the results of their investigations form the foundation of all the measurement systems we use in audio today. The folks at Bell Laboratories get the credit for finding out how we judge sound power, how quiet a sound an average person can hear, and almost all of the many other details about sound you must know before you can work with it successfully.

From this basic research, Bell Labs developed a system of units that could be applied to all phases of the system. Sound traveling on wires as electrical energy, sound on tape as magnetic energy, sound in air; anyplace that sound is, or has been stored as energy until some future time when it will again be sound, can be described by using the human ear-related system of numbers called "bels" in honor of Alexander Graham Bell, the inventor of the telephone.

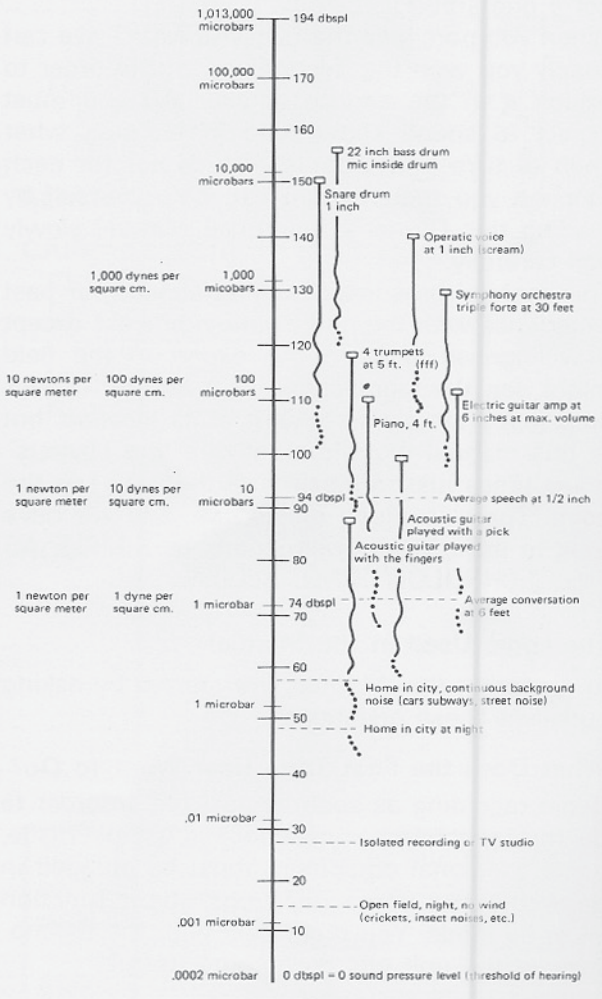
What is a bel and what does it stand for?

It means, very simply, twice as loud to the human ear. Twice as loud as what? An obvious question. The bel is always a comparison between two things. No matter what system of units of measure you are working with at the time, you must always state a value as a reference before you can compare another value to it by using bels, volts, dynes, webers—it doesn't matter, a bel, or ear-related statement of "twice as loud" is always a ratio, not an absolute number. Unless a zero, or "no difference" point is placed somewhere, no comparison is possible.

There are many positive and definite statements of reference in use today. But before we go over them, we should divide the "bel" into smaller units. "Twice as loud" will be a little crude to be used all the time. How about one tenth of a bel? Okay, the decibel it is, and 0 means "no difference, same as the reference". It seldom means "nothing". Now, if you double the power, is that twice as loud? No, it is only 3 dB more sound. If you double an electrical voltage, is it twice as loud? No, it is only 6 dB more sound. The unit quantities must follow nonlinear progressions to satisfy the ears' demand.

Remember, decibels follow the ears. All other quantities of measure must be increased in whatever units are necessary to satisfy the human requirements, and may not be easy to visualize. Sound in air, our beginning reference, is the least sound the human ear (young men) can detect at 1000 to 4000 Hz. Bell Labs measured this value to be .0002 microbar, so we say 0 dB = .0002 microbars and work our way up from the bottom, or from the point at which there is "no perceivable sound to humans". Here is a chart of sounds and their ratings in dB, using .0002 microbar pressure change in air as our reference for 0 dB spl (Sound Pressure Level).

SOUND AND MUSIC REFERENCE



Since the reference is assumed to be the lowest possible audible value, dB spl (sound pressure level) is almost always positive, and correctly written should have a + sign in front of the number. But it is frequently omitted. Negative dB spl would indicate so low an energy value as to be of interest to a scientist trying to record one cricket at 1,000 yds. distance, and is of no significance to the multichannel recordist. Far more to the point is the question "What is a microbar?" It is a unit of measurement related to atmospheric pressure and although it is extremely small, it must be divided down quite a lot before it will indicate the minimum pressure change in air that we consider minimum audible sound. This will give you a better idea of the sensitivity of the human ear.

One whole atmosphere, 14.70 pounds per square inch, equals 1.01325 bars. So one whole atmosphere in microbars comes out to be 1,013,250. One microbar of pressure change is slightly less than one millionth of an atmosphere, and you can find it on our chart as 74 dB spl. It is not terribly loud, but it is certainly not hard to hear. As a matter of fact, it represents the average power of conversational spech at 6 feet. This level is also used by the phone company to define normal earpiece volume on a standard telephone. Now think about that minimum audible threshold again:

.0002 microbar.

That's two ten thousandths of a millionth part of one atmosphere!

This breakdown of one reference is not given just to amaze you, or even to provide a feel for the quantity of power that moderate levels of sound represent. Rather it is intended to explain the reason we are saddled with a ratio/logarithm measurement system for audio. Adding and subtracting multi-digit numbers might be easy in this age of pocket calculators, but in the 1920's when the phone company began its research into sound and the human ear, a more easily handled system of numbers became an absolute necessity. Convenience for the scientist and practical engineer, however, has left us with a system that requires a great deal of complex explanation before you can read and correctly interpret a "spec sheet" for almost any piece of gear.

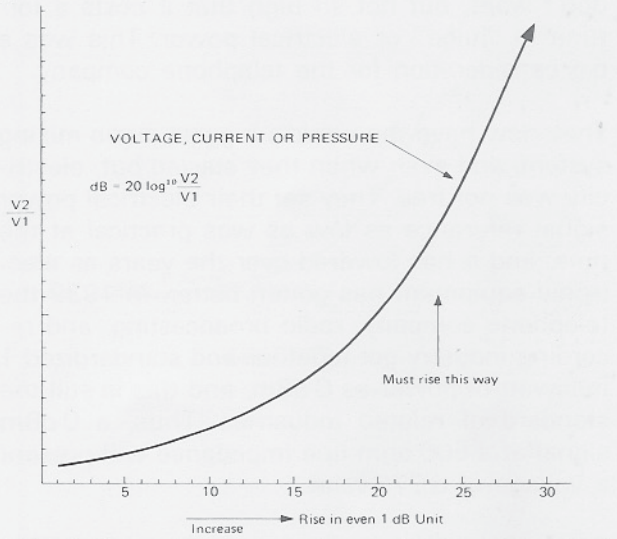
Here are the formulae for unit increment, but they are necessary only for designers. And unless you build your own gear, you won't have to deal with them. For power (watts) increase or loss, calculate by the following equation:

$$10 \ LOG_{10} \frac{P2}{P1} = N\,(dB)$$

For voltage, current or pressure calculations:

$$20 \ LOG_{10} \frac{V2}{V1} = N\,(dB)$$

Plotting the points resultant from using these equations we come up with the following chart. Once we have this chart, we can see the difference between the way humans perceive sound and the amount of force it takes to change air pressure. Unfortunately, the result is not a simple "twice as much pressure" of sound to be heard as "twice as loud". If you plot decibels as the even divisions on a graph, the unit increase you need is a very funny curve.



This is how the ear works, and we must adapt our system to it. We have no choice if we expect our loudspeaker to produce a sound that resembles the original sound we begin with. The high sensitivity to sound of the human ear produces a strong "energy" illusion that has confused listeners since early times. How powerful are the loudest sounds of music in real power? Can sound be used as a source of energy to do useful work, such as operating a car? For any normally "loud" sound the answer is, regrettably, no! Perhaps not so regrettably, consider what would happen if one pound of pressure was applied not to your head, but directly to your inner ear. One pound of air pressure variation is 170 dB spl! This amount of "power" might do some useful work—but not much, it's still only one pound and to make use of it you will have to stand one mile away or you will go deaf immediately.

If we reduce our sound power to realistic musical values, we will not be injured, but we will have almost nothing (in real power terms) to run the mic with! This low available energy is the reason that high gain amplifiers are required for microphones.

When we take a microphone and "pick up" the sound, we do have some leeway in deciding how much energy we must have in order to operate the electrical part of our system. If we can decide that we don't have to truly hear the signal while we are processing it from point to point and we can wait until the electronic devices have done all their routing and switching before we need audible sound, we can lower the power of the signal. What is good value for a reference here? Well, we need to have enough energy so that the signal is not obscured by hiss, hum, buzz or other unpleasant things we don't want, but not so high that it costs a fortune in "juice" or electrical power. This was a big consideration for the telephone company.

They now have the world's biggest audio mixing system, and even when they started out, electricity was not free. They set their electrical power signal reference as low as was practical at the time, and it has lowered over the years as electronic equipment has gotten better. In 1939 the telephone company, radio broadcasting, and recording industry got together and standardized 1 milliwatt of power as 0 dBm, and this is still the standard of related industries. Thus, a 0 dBm signal at a 600 ohm line impedance will present a voltage of 0.775 volts.

Once again, we owe you an explanation. Why does it say *zero* on the meter? What is an ohm? Why 600 of them and not some other value? What's a volt? Let's look at one thing at a time.

1. The logic of ZERO on the meter is another hangover from the telephone company practice. When you start a phone call in California, the significant information to a telephone company technician in Boston is—did the signal level drop? If so, how much? When the meter says ZERO it indicates (to the phone company) that there has been no loss in the transmission, and all is well. The reference level is one milli-watt of power, but the gain or loss is in the information the meter was supposed to display, so the logic of ZERO made good sense, and that's what they put on the dial. We still use it even though it's not logical for anything else, and the idea of a reference level described as a "no loss" ZERO, no matter what actual power is being measured is so firmly set in the minds of everyone in the audio world that it is probably never going to change.

2. One ohm is a unit of resistance to the passage of electrical energy. The exact reasons for the choice of 600 ohms as a standard are connected to the demands of the circuits used for long distance transmission and are not simple or easy to explain. Suffice it to say that the worst possible thing you can do to a piece of electronic equipment is to lower the resistance it is expected to work into (the load). The lower the number of ohms, the harder it is to design a stable circuit. When you think about "load", the truth is just the opposite of what you might expect! 0 ohms is a "short circuit", no resistance to the passage of signal. If this condition occurs before your signal gets from California to Boston, you won't be able to talk—the circuit didn't "get there", it "shorted out". Once again, telephone company logic has entered the language on a permanent basis. Unless the value for ohms is infinity (no contact, no possible energy flow) you will be better off with a higher value, and many working electronic devices have input numbers in the millions or billions of ohms.

3. A volt is a unit of electrical pressure and by itself is not enough to describe the electrical power available. To give you an analogy—that may help, you can think of water in a hose. The pressure is not the amount of water, and fast flow will depend upon the size of the hose (impedance or resistance) as well. Increase the size of the pipe (lower the resistance, or Z) and pressure (volts) will drop unless you make more water (current) available to keep up the demand. This analogy works fairly well for DC current and voltage, but alternating current asks you to imagine the water running in and out of the nozzle at whatever frequency your "circuit" is working at, and is harder to use a mental aid. Water has never been known to flow out of a pipe at 10,000 cycles per second.

This reference level for a starting point has been used by radio, television, and many other groups in audio because the telephone company was the largest buyer for audio equipment. Most of the companies that built the gear started out

working for the phone company and new audio industries, as they came along, found it economical to use as much of the ready-to-hand stuff as they could, even though they were not routing signals from one end of the world to the other.

Must we use this telephone standard for recording? Its use in audio has been so widespread that many people have assumed that it was the only choice for quality audio. Not so.

A 600 ohm, 3-wire transformer-isolated circuit is a necessity for the telephone company, but the primary reason it is used has nothing to do with audio quality. It is noise, hum and buzz rejection in really long line operation (hundreds and hundreds of miles).

Quality audio does not demand 600 ohm, 3-wire circuitry. In fact, when shielding and isolation are not the major consideration, there are big advantages in using the 2-wire system that go well beyond cost reduction. It is, as a system, inherently capable of much better performance than 3-wire transformer-isolated circuits.

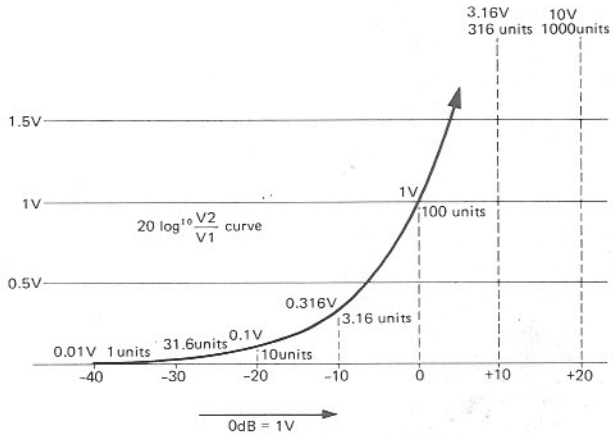
We think that the 2-wire system is a wise choice for our PORTASTUDIO. The internationally accepted standard (IEC) for electronics of this kind used a voltage reference without specifying the exact load it is expected to drive. The reference is this:

$$0 = 1 \text{ volt}$$

This is now the preferred reference for all electronic work except for the telephone company and some parts of the radio and television business. Long distance electronic transmission still is in need of the 600-ohm standard.

If your test gear has provision for inserting a 600-ohm load, be sure the load is not used when working on TASCAM equipment.

Now that we have given a reference for out "0" point, we can print the funny curve again, with numbers on it, and you can read voltages to go along with the changes in dB.



IMPEDANCE MATCHING AND LINE LEVELS

All electronic parts, including cables and non-powered devices (mics, passive mixers and such), have impedance, measurable in ohms (symbol Ω or Z). Impedance is the total opposition a part presents to the flow of signal, and it's important to understand some things about this value when you are making connections in your mixing system. The outputs of circuits have an impedance rating and so do inputs. What's good? What values are best? It depends on the direction of signal flow, and in theory, it looks like this:

OUTPUTS —— **Plug Into** ——▶ **INPUTS**

It is generally said that the output impedance (Z) should be as low as possible. 100 ohms, 10 ohms. The lower, the better, in theory. A circuit with a low output impedance will offer a low resistance to the passage of signal, and thus will be able to supply many multiple connections without a loss in performance or a voltage drop in any part of the total signal pathway. Low impedance values can be achieved economically by using transistors and integrated circuits, but other considerations are still a problem in practice, such as:

1. The practical power supply is not infinitely large. At some point, even if the circuit is capable of supplying more energy you will run out of "juice".

2. Long before this happens, you may burn out other parts of the circuit. The output impedance may be close to the theoretically ideal "ohms" but many parts in the practical circuit are not. Passing energy through a resistance generates heat and too much current will literally burn parts right off the circuit card it steps are not taken to prevent catastrophic failure.

3. Even if the circuit does not destroy itself, too high a demand for current may seriously affect the quality of the audio. Distortion will rise, frequency response will suffer, and you will get poor results.

Inputs should have very high impedance numbers, as high as possible (100,000 ohms, 1 million ohms, more, if it can be arranged).

A high resistance to the flow of signal at first sounds bad, but you are not going to build the gear. If the designer tells you his input will work properly and has no need for a large amount of signal, you can assume that he means what he says. For you, a high input impedance is an unalloyed virtue. It means that the circuit will do its job with a minimum of electrical energy as a beginning. The most "economical" electronic devices in use today have input impedances of many millions of ohms, test gear for example, voltmeters of good quality must not draw signal away from what they are measuring, or they will disturb the proper operation of the circuit. A design engineer needs to see what is going on in his design without destroying it, so he must have an "efficient" device to measure with.

SOURCE (Output) —— **Plugs into** ——▶ **LOAD (Input)**

The classic procedure for measuring output impedance is to reduce the load's impedance until the output voltage drops 6 dB (half the original power) and note what the load value is. In theory, you now have a load impedance that is equal to the output impedance. If you gradually reduce the load (increase the input impedance), the dB reading will return slowly to its original value. How much drop is acceptable? What load will be left when an acceptable drop is read on the meter?

Traditionally, when the load value (input Z) is approximately seven times the output impedance, the needle is still a little more than 1 dB lower than the original reading.

Most technicians say, "1 dB, not bad, that's acceptable." We at TASCAM must say that we do not agree. We think that a seven-to-one ratio of input (7) to output (1) is not a high enough ratio, and here's why:

1. The measurement is usually made at a mid-range frequency and does not show true loss at the frequency extremes. What about the drop at 20 Hz or 30 kHz?

2. All outputs are not measured at the same time. Most people don't have twenty meters, we do. Remember, everybody plays together when you record and the circuit demands, in practice, are simultaneous. All draw power at the same time.

Because of this widely misunderstood rule of thumb—the seven-to-one ratio, we will give you the values for output impedance.

True Output Impedance
Even though the true output impedance may be low, say 100 ohms, it takes a lab to check the rule of thumb, so for the practical reasons we have explained, the use of the ratio method of impedance calculation must be changed to a higher ratio. We prefer 100:1 if possible and we consider 50:1 to be the minimum ratio that we think safe. Because of this, we will give you a number for ohms that you can match, Minimum Load Impedance. No calculations, we have made them already.

Minimum Load Impedance
MAKE CERTAIN THAT YOU CONNECT NO TOTAL LOAD IMPEDANCE LOWER (numerically) THAN THESE FIGURES.

Access Send Output	1.7k ohms
Tape Output	1.7k ohms
Aux Send Output	1.7k ohms
Tape Cue Output	1.7k ohms
Line Output	1.7k ohms
Aux Output	1.7k ohms

Nominal Load Impedance
Our specifications usually show 10,000 ohms as a Nominal Load Impedance. This load will assure optimum performance. Remember, any impedance lower than 10,000 ohms is more load.

Input Impedance
Input impedance is more straightforward and requires only one number. Here are the values for the PORTASTUDIO:

Mic/Line Input	60k ohms
Access Receive Input	68k ohms
Aux Receive Input	37k ohms

If one output is to be "Y" connected to two inputs the total impedance of the two inputs must not be lower than the minimum load impedance, mentioned above, and if it becomes necessary to increase the number of inputs with slight reduction of the load specifications, you must check for a drop in level, a loss of headroom, low frequency response, or else suffer from a bad recording. If one input is 10,000 ohms, another of the same 10,000 ohms will give you a total input impedance (load) of 5,000 ohms. To avoid calculations you can do the following when you have two inputs to connect to one output.

Take the lower value of the two input impedances and divide it in half. If the number you have is greater than the minimum load impedance, you can connect both at the same time. Remember, we are not using the true output impedance we are using the adjusted number, the minimum, output load impedance.

If you must have exact values here is the formula for dissimilar 2 loads or inputs:

$$RX = \frac{R1 \times R2}{R1 + R2}$$

When you have more than two loads (inputs), just dividing the lowest impedance by the number of inputs will not be accurate unless they are all the same size. But if you still get a safe load then the minimum load impedance by this method, you can connect without worry.

If you must have exact values, here is the formula for more than 2 loads or inputs:

$$RX = \frac{1}{\frac{1}{R1} + \frac{1}{R2} + \frac{1}{R3} + \cdots + \frac{1}{Rn}}$$

RX = Value of Total Load

Finding Impedance Values on Other Brands of Equipment

When you are reading an output impedance specification, you will occasionally see this kind of statement:

Minimum load impedance = X ohms

or

Maximum load impedance = X ohms

These two statements are trying to say the same thing, and can be very confusing. The minumum load impedance says: please don't make the NUMBER of ohms you connect to this output any lower than X ohms. That's the lowest NUMBER. The second statement changes the logic, but says the exact same thing.

Maximum load impedance refers to the idea of the LOAD instead of the number, and says: please don't make the LOAD any heavier. How do yo increase the load? Make the number lower for ohms. Maximum load means minimum ohms, so read carefully.

When the minimum/maximum statement is made, you can safely assume that the manufacturer has already done his calculations, and the number given in ohms does not have to be multiplied. You can MATCH the value of your input to this number of ohms successfully; but as always, higher ohms will be okay (less load).

Occasionally, a manufacturer will want to show you that 7 times the output Z is not quite the right idea and will give the output impedance and the correct load this way, they will call the output impedance the True Output Impedance and then will give the recommended minimum LOAD impedance. It may be a higher or lower ratio than 7 times and will be whatever the specific circuit in question requires.

REFERENCE LEVELS

We should talk about one more reference, a practical one.

Anyone who has ever watched a VU meter bounce around while recording knows that "real sound" is not a fixed value of energy. It varies with time and can range from "no reading" to "good grief" in less time than it takes to blink. In order to give you the numbers for gain, headroom and noise in the PORTASTUDIO, we must use a steady signal that will not jump around. We use a tone of 1000 cycles and start it out at a level of −60 dB at the mic input, our beginning reference level. All levels after the mic input will be higher than this, showing that they have been amplified, and eventually we will come to the last output of the PORTASTUDIO—the line-out and the reference signal there will be −10 dB, our "line level" reference.

From this you can see that if your sound is louder than 94 dB spl, or your mic will produce more electricity from a sound of 94 dB spl than −60 dB, all these numbers will be changed. We have set this reference for mic level fairly low. If you examine the sound power or sound pressure level (spl) chart on page 6 you will see that most musical instruments are louder on the average than 94 dB spl, and most commercial mics will produce more electricity than the −60 dB for a sound pressure of 94 dB, so you should have no problems getting up to "O VU" on your recorder.

We should also make a point of mentioning that the maximum number on this chart represents "peak power" and not average power. The reason? Consider if even some momentary part of your recording is distorted, it will force a re-recording and it is wisest to be prepared for the highest values and pressure even if they only happen "once in a while". On this point, statistics are not going to be useful, the average sound pressure is not the whole story. The words themselves can be used as an example. Say the word "statistics" close to the mic while watching the meters and the peak LED level detector. Then say the word "average". What you are likely to see are two good examples of the problems encountered in the "real world" of recording. The strong peaks in the "s" and "t" sounds will probably cause the LED's to flash long before the VU meter reads anywhere near "zero" while the vowel sounds that make up the word "average" will cause no such drastic action.

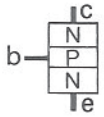
To allow peaks to pass undistorted through a chain of audio parts, the individual gain stages must all have a large reserve capability. If the average is X then X + 20 dB is usually safe for speech, but extremely percussive sounds may require as much as 40 dB of "reserve" to insure good results. Woodblocks, castanets, latin percussion (guido, afuche) are good examples of this short term violence that will show a large difference between "LED flash" and actual meter movement. When you are dealing with this kind of sound, believe the LED, it is telling you the truth.

If you are going to record very loud sounds you may produce more electrical power from the mic than the PORTASTUDIO can handle as an input. How can you estimate this in advance? Well, the spl chart and the mic sensitivity are tied together on a one-to-one basis. If 94 dB spl gives −60 dB (1 mV) out, 104 dB spl will give you −50 dB out, and so forth. Use the number, on our chart for sound power together with your mic sensitivity ratings to find out how much level, then check that against the maximum input levels for the various jacks on the PORTASTUDIO. If your mic is in fact producing −10 dB or line level, there is nothing wrong with plugging it into the line level connections on the mixer. You will need an adaptor, but after that it will work!

Most mic manufacturers give the output of their mics as a minus-so-many-dB number, but they don't give the loudness of the test sound in dB, it's stated as a pressure reference (usually 10 microbars of pressure). This reference can be found on our sound chart. It is 94 dB spl, 10 microbars, 10 dynes per cm^2 or 1 Newton per square meter. For mics, the reference "0" is 1 volt (dB). So, if the sound is 94 dB spl, the electrical output of the mic is given as −60 dB, meaning so many dB less than the reference 0 = 1 volt. In practice, you will see levels of −60 dB for low level dynamics, up to about −40 dB or slightly higher for the better grade of condenser mics available today. TASCAM recorders and mixers work at a level of −10 dB reference to 1 volt (.316 volt) so, for 94 dB spl, a mic with a reference output of −60 dB will need 50 dB of amplification from your PORTASTUDIO or recorder in order to see "O VU" (−10 dB) on your meter. Now, if the sound you want to record is louder than 94 dB spl, the output from the mic will be more powerful and you will need less amplification from your PORTASTUDIO to make the needles on your recorder read "O VU".

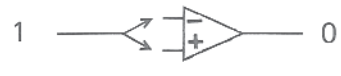
HOW AMPLIFIERS WORK

Amplifiers do not acually pass signal electrons to their outputs, rather, the signal is used to control a junction of semiconductor material. This junction is neither truly a conductor, (such as copper) or a non-conductor (such as glass), the changes in signal energy alter the junction and cause it to valve or control the power generated by the unit power supply. The extra energy must come from somewhere, so you must have a battery, or a wall plug to provide a source. When you amplify a signal you don't "pass it on" you "duplicate it" at a higher power, the simplest semiconductor amplifier is the transistor, a sandwich of three layers of differing semiconductor materials.

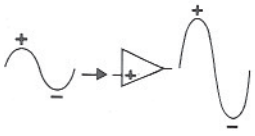


When signal is applied to Pin b it causes the junction of Pin e to conduct, and a replica of the input signal applied to Pin b appears on Pin c. The amount of enlargement of the energy depends on many factors, The transistor type, the power available from the supply, and the signal conditions. Many other parts are usually employed to insure that the replica of the signal is reasonably close to the original. In the PORTASTUDIO we use many pre-packaged groups of parts called integrated circuits, or IC's.
Integrated circuits come in all kinds of configurations, computer control, timers, on-off transistor "switches" and many more. For use in audio the most common configuration is called the Operational Amplifier. The name relates to its original use in computers as a mathematical modeling device (it will "add" and "subtract" signals and can be used as a calculating stage). But it has found wide acceptance in audio applications as well.
This "package" has more than one signal input. One is "positive", turning the supply power up and down "in step" with the signal variation. The other is wired internally to control the supply in reverse. It valves backwards and says to the amplifier, don't instead of do. Its called a "negative" input. Here are three common ways to use the inputs on an operational amplifier.

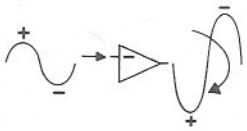
1. If the same signal appears on both pins, the output of the amp is zero.



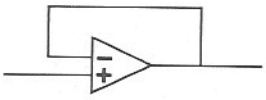
2. If a small signal appears at the positive input, the "gain" can be as much as 10,000 or more (100 dB) and the output is "in step with the input."



3. If a small signal is applied to the negative or "no you don't" input the gain is again potentially large, but when signal is positive, the output is negative. In musical terms, when the drum head moves foward, the loudspeaker will be forced to move backwards, and that's not what we want. We must make sure that we don't leave this "out of phase" condition uncorrected if we use a negative input in our design.

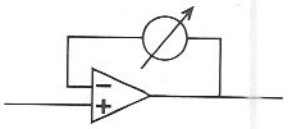


4. A negative input can be used as a gain controlling "port" and is almost always connected to its own output! Like this:



Contrary to what you might think, this configuration is widely used! Whatever goes in, comes out with no change in level. Not nothing, mind you, but unity, the same. Now if we don't connect all of the output back to the negative "don't" port, but instead we place a control in the reverse line, we can change the "gain" of the amplifier by simply rotating the control. We won't need a soldering iron and a bag of parts. This control will allow us to adjust the gain to suit the needs of the moment, a lot or a little as needed. The input signal says "go up all the way" and our reverse control says "well not quite so far as all that" and you have control. The reason that it works is that first paragraph. SIGNAL DOES NOT ACTUALLY FLOW THROUGH AN AMP. It simply controls the "Valving" action and the total result of the two commands sets the final size of the power supply duplicate of the original input.

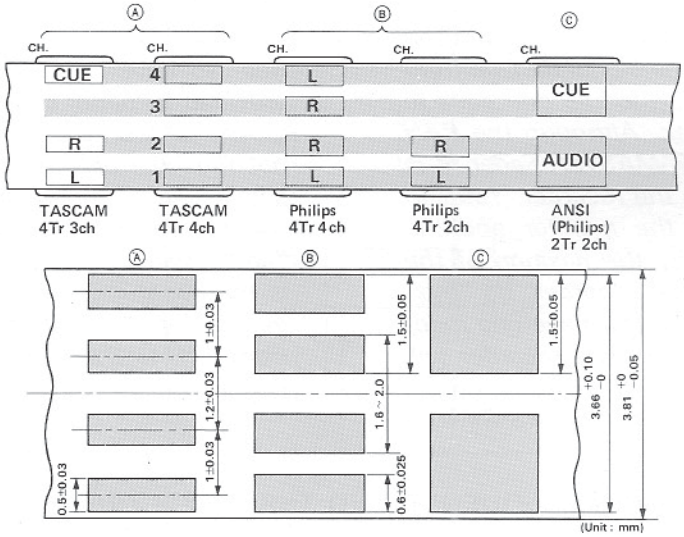
In this unit, the maximum result or gain when the TRIM control is rotated fully rightwards is 50 dB. The minimum gain when the control is rotated fully to the left is 0 dB.



This configuration is usually referred to as a Feedback control. In the P.A. business, such a term is not considered a benefit, it always refers to an unpleasantness that can stop the show. In the world of semiconductors its not without merit, but it does requre intelligent use. Because of this reverse, or Feedback control aspect of the trim, we consider it unwise to adjust while signal is being recorded. Obviously, you must adjust when signal is present, but the possible negative side effects on amp stability and distortion indicate that any use of this control should be a "set and leave alone" kind of action.

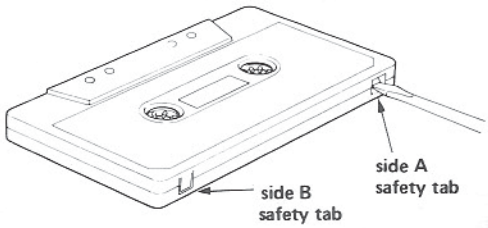
If you are working alone, you will have both hands on your instrument when a "take is in progress, so the use of this control is not completely ruled out as a method of signal management.

1. **The PORTASTUDIO uses a "base" speed of 3-3/4 IPS and a discrete 4 channel TASCAM format head.**
   It will not play the standard stereo format. You will have to mix to another deck to get a playable 2 track tape. Conversely, the cassettes you already have will not reproduce properly on the PORTASTUDIO because of the headstack and base speed differences.



2. **Tapes made on the TASCAM models 133 and 144 with Dolby B noise reduction are not compatiable with the Model 244 DBX II and will not give satisfactory results.**
   A careful examination of the headstack layout will show that 3 models do use the same head, and that the sequence of tracks is inverted on the 144. This inversion is significant to the audio visual user who must reproduce a show control tape on a variety of formats, but the fact that the 144 and 244 noise reduction systems are not compatible and CANNOT BE BYPASSED is far more significant, and the reason for this warning. Tapes made on one unit will not be damaged by playing on another, but the sonic results will

   noise reduction systems.
3. **The use of C-120 cassettes is not recommended.**
   The 1/2 mil tape thickness of the C-120 can lead to winding troubles, crimping, wrinkling and other damage to the oxide coating which will destroy your work. Consider this: it is not unusual to play the tape 100 times before you are ready to mix down, so select a cassette "time" that relates as closely as possible to the length of the program you plan on doing. Consider also what will happen if you wish to protect your "master" by punching out the record protect tabs. A short selection on a long cassette will force you to waste the usused portion of tape. Shortest is best. The shorter cassettes use the thickest and most durable tape. Remember that the 2x normal speed and the "one side only" 4 track, one-way format mean that a C-90 cassette will run for only 22-1/2 minutes, a C-60 for 15 minutes a C-45 for 11-1/4 minutes and a C-30 for 7-1/2 minutes.
4. **To protect a finished master. It is necessary to punch out both "protect" tabs.**
   Even though you are recording in only one direction, you are using the entire width of the tape. If you only knock out one tab, you could still put the tape back in "upside down" and erase your master.



5. **The PORTASTUDIO record circuits cannot be "set" for more than one tape type externally.**
   The PORTASTUDIO is internally adjusted for gamma-ferric oxide high bias tapes (70 micro-sec. EQ) such as TDK-SA, Maxell UDXL-II, or the equivalent, we suggest that you select one brand and use it exclusively. Doing this will allow you to work without having to compensate for the minor differences in performance between brands.

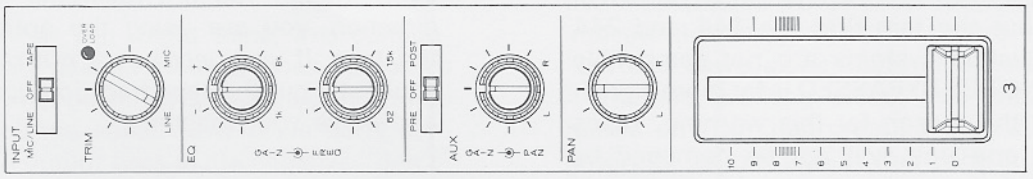
SPECIAL CAUTION FOR THE FOUR CHANNEL RECORD FEATURE

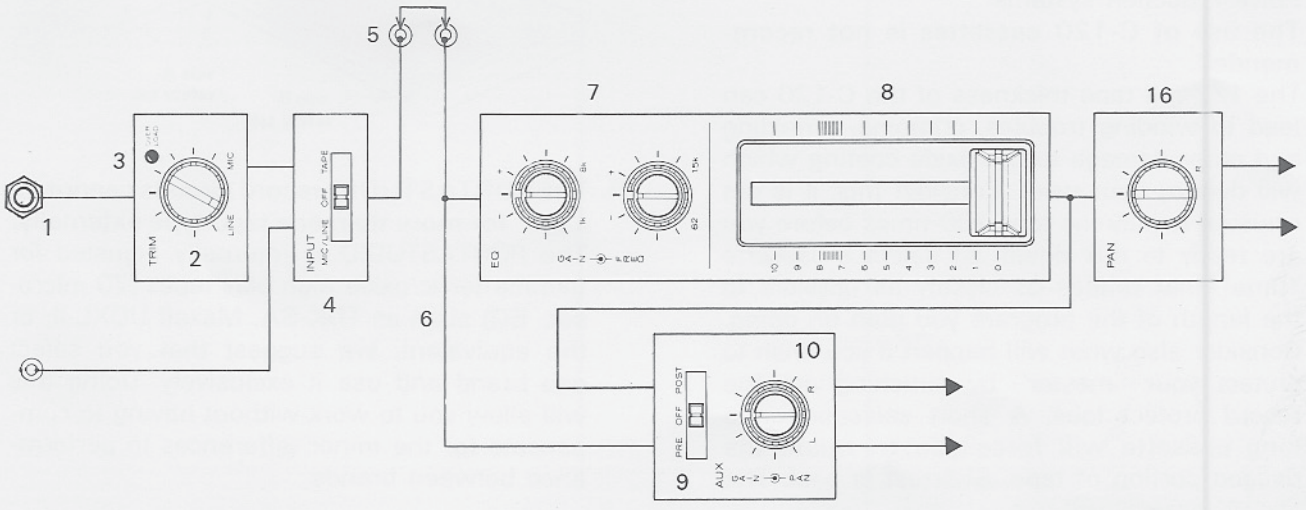
Four channel record special function and meter channel record logic

When the Four Channel Record function is selected, each input module is assigned to its corresponding track on the recorder, and the four meters now read the signal derived from this point in the wiring; module one goes to meter one, and so forth. Although the PAN Pots still feed the L+R MASTER Fader, it is no longer connected to the recorder. You can still hear "stereo" in the monitor and the level will be affected by the position of the L+R MASTER but to the recorder, no "mixing" is possible. You will now have one module feeding each track, and the recorder will ALWAYS EARASE ALL FOUR TRACKS PRIOR TO RECORDING ALL FOUR TRACKS SIMULTANEOUSLY! YOU CANNOT MAKE AN OVERDUB IN THIS MODE. The Tape Cue circuit is disabled, depriving you of the necessary headphone feed, and anything that is already on the tape will be erased when you start 4 Ch Rec, so take care. This special function is provided to allow copies to be made, and to record all four tracks at once. It is definitely not the "normal" method of use. True, it is possible to begin a tape in this mode, you can record one, two or three tracks at one "pass" with this logic, switch to SYNC (That is the overdub mode) and finish the remaining track or tracks but if you forget to switch from 4CH to SYNC, YOU WILL ERASE YOUR FIRST "PASS".

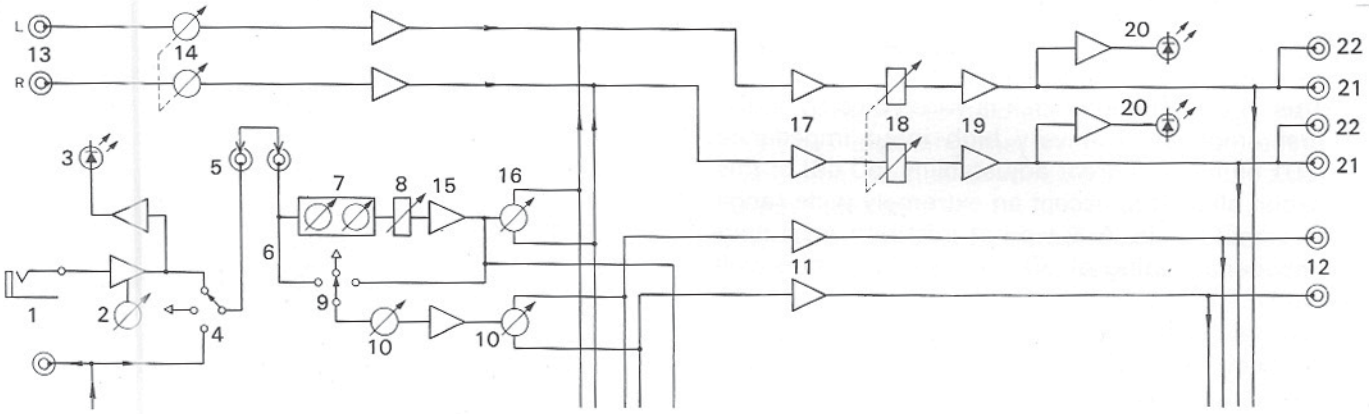
UNDERSTANDING THE WIRING SEQUENCE AND THE BLOCK DIAGRAM

In most instances, the physical arrangement of the controls on the top panel has very little to do with the sequence of electronic parts inside. The actual "wiring order" will determine what goes where and how, so this is the information you need in order to use the PORTASTUDIO successfully. As an example, if the controls on an input module were actually placed in the order that they were "wired up" inside, the top panel would look like this. We'll include the patch points from the back panel in their wiring order as well as the faders and switches. Here's the comparison!





While this arrangement of controls might help the beginner to understand the flow of signal in the module, it would be very inconvenient to operate. Still, the wiring sequence must be understood before the more complex functions of the PORTASTUDIO can be used, so our description of the system will follow the block diagram, or wiring sequence.



This drawing shows all the controls, switches, amplifying stages and connectors in their actual "as wired" sequence. Learning to read it will provide the answers to any question concerning "what comes where" on the inside. Things like "does the accessory send-receive jack pair appear before or after the equalizer?" can be answered quickly. (The jacks are connected before the EQ section, so you will not have EQ on any signal derived from this point.)

It you have no prior experience in the reading of block diagrams, you can use the three illustrations we have provided here as a translator. Compare the reconstructed (as wired) module drawing with the symbol used on the block to find out what each graphic represents.

In multitrack recording systems, a description of signal "sequence" or flow is made more confusing by the multiple uses of the same circuit, depending on what point in the process you are considering. Obviously, there is no real need to duplicate functions that can use the same parts at a later time in the process. For example, you don't need a separate EQ section for the Mic/Line In and Tape In functions, a simple re-routing switch to select the current "start" signal, Mic/Line, or Tape (remix) will be all that is necessary. If we were to follow the layout of the top panel, this description of controls on the PORTASTUDIO would have to explain an extremely large number of different concepts with no certainty that your working "method" would be one of the ones covered.

To save space, we will stick to the wiring sequence, and describe the "options" as we come to them on a "once through" basis, and leave the practical decisions about a specific "working method" to you, the operator. This "confusion" is an unavoidable side effect of flexibility, but a few weeks of operation will make you more familiar with the PORTASTUDIO. It is not necessary to absorb all the possible routings in the first ten minutes of operation.

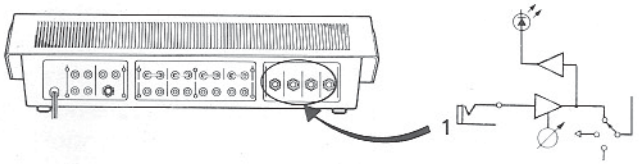
Before you begin reading the next section of this manual, flip out the double fold on the back cover. On this page we have printed a complete block diagram for the PORTASTUDIO. The usual method of preparing a block diagram is to draw one example of each circuit and then say "how many" there are of each type. For this "block" we think it wiser not to use any repeats. Each section is drawn out completely.

When printed in a complete form, a system block diagram can look pretty complex. To aid you in your initial understanding we will continue to use a smaller two step version of the translation system shown here. In the Parts List section you will see;

1. The physical location on the PORTASTUDIO of the part being discussed.
2. The symbol for the part and its location on the Block Diagram.

1. MIC/LINE INPUT Connector

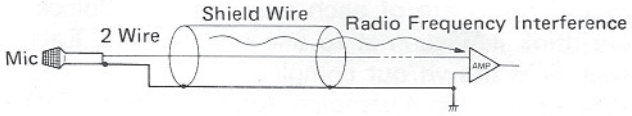
There are 4 of these, one for each Mixer Input section.



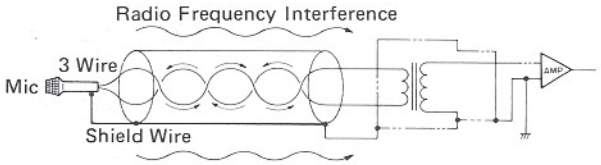
This 1/4 inch phone jack is wired directly to the first amplifier. The very high input impedance (60k ohms) and great adjustability (50 dB) of this circuit allow it to accept an extremely wide range of signal levels. Any type of mic with an output impedance rating of 50 ohms to 10k ohms will work.

Any line level source below 0.4V may also be connected here.This circuit will also accept the output of electric instrument inductive pickups (guitar, electric bass) without the need for a "direct box" or transformer, but such devices may be used if desired. The circuit is extremely adaptable. Why use a transformer if it is not absolutely necessary? There may be a good reason, and it applies primarily to mics used with cable runs exceeding 10 feet. The low power signal that mics (and some instrument pickups) generate must frequently be protected and isolated from other low power signals in the real world. Radio, power line hum, buzz, crackles and switching noise when motors start up (do you have an air conditioner on your AC line?)—all these unwanted signals must be kept out of the very high gain amplifiers that are needed to raise the mic signal to a working level. The balanced or three-wire mic circuit and input isolation transformer become the only sure way to deal with the problem.

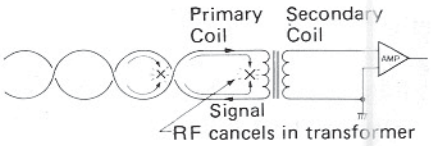
Here's how it works:



Any signal will pass to amplifier, no rejection.

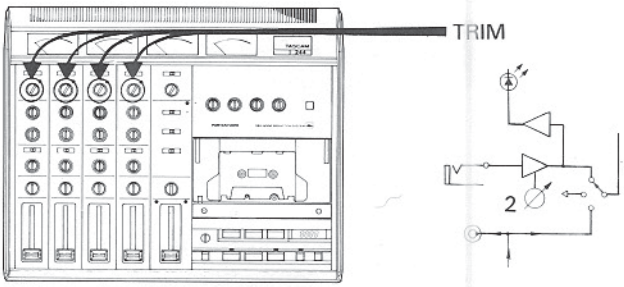


Audio signals from mic have opposite polarity. Buzz, hum, and RFI have common polarity.



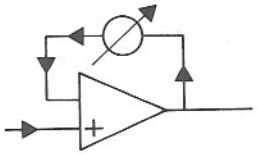
Signals with opposite polarity in the primary coil will generate current in the secondary coil. Signals with common polarity will cancel out in the primary coil and will not pass to the secondary coil. No signal in the secondary coil means no signal in the amplifier. You leave the "junk" behind and get only the mic signal.

2. TRIM Control



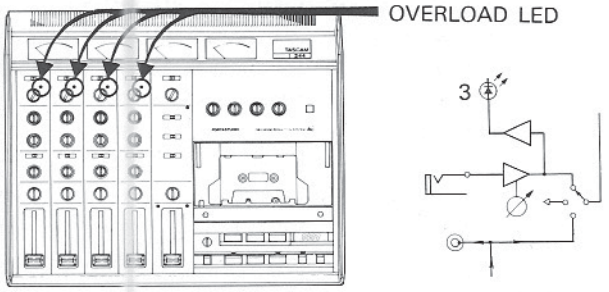
The next symbol you will see on the block diagram signal "line" is a triangle, pointing "down the line". This is the engineering symbol for an amplifier, or gain stage. The "point" of the triangle represents the output and indicates the direction of signal flow. Knowing that there is an "amp" in a specific location is not just an academic curiosity. When you are trying to optimize the performance of a multitrack mixer and it has more than one possible "route" to get your job done, the pathway to choose is always the one that shows the fewest number of amplifying stages. No matter how "good" an amp is, it will always add some additional noise and distortion to your signal. The smaller number of amps will always do less injury to your sound. Amplifiers don't show on the outside of the unit, but the controls that operate them do, so we head this paragraph with the name of the control. This control will alter the gain of the first amplifier in the mixer. It will effect the level of Mic/Line signal, but has no effect on the TAPE (remix) position of the Input select switch. If you trace the line from the switch back to the recorder section, you will see that Tape signal has been amplified already, and does not need to pass through the Mic/Line amp. When less is more, we go for it. With TRIM rotated fully leftwards, the gain of the first amp is 0 dB. when rotated fully rightwards, the gain is 50 dB. Maximum input signal to this first gain stage is +15 dB (5.6 V) at minimum gain (full left setting). At max (full right setting), −35 dB (13 mV).

This TRIM controls the gain of the amplifier by adjusting the amount of output signal returned to a secondary input "Port" (Feedback Control). If you want to know more about how amplifiers are controlled, read the section How Amplifiers Work on page 13.



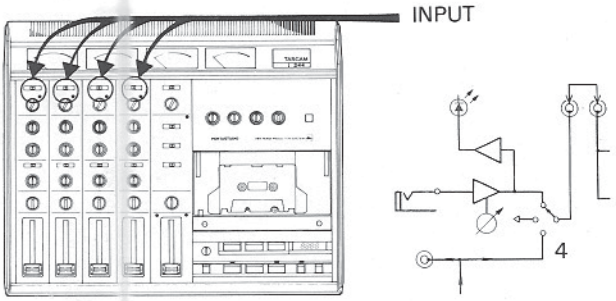
Because of this feedback control, we consider it unwise to adjust this TRIM while signal is being recorded. Obviously, you must adjust when signal is present, but when serious recording is in progress the possible negative side effects on amplifier stability and distortion indicate that you should "mix" with the straight line faders *only* and adjust this special TRIM during the rehearsals.

3. OVERLOAD LED



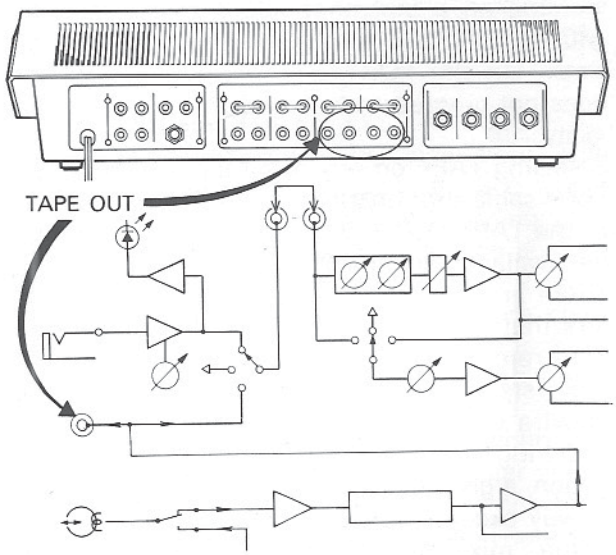
When signals high enough to make the ACCESS-SEND jack output exceed +12 dB are applied to the MIC/LINE INPUT Connector, this LED will light. Adjust the TRIM control until this LED remains out when signal is present. When working with extremely percussive transient material, it may require full negative Trim to prevent this LED from flashing on strong "peaks". Changing to a less "sensitive" mic may also be necessary.

4. INPUT Select Switch



This switch has three positions.

- Set Left: Selects the MIC/LINE INPUT Connector on the rear panel of the PORTASTUDIO.
- Set Center: Acts as a "mute".
  This "mute" can be useful in many ways. When used on Mic/Line signals it will allow you to turn on a signal without having to move the fader to a preset "mark" accurately. This "drop in" function with all controls preset can be used to edit out undersirable sections from a track when you are remixing. Prior to your final mix, the use of this mute function will allow you to hold all your preliminary mix settings including the level set by the fader, and still silence a module while you "fine tune" another.
- Set Right: (TAPE) Selects an internal connection to the recorder's playback amplifiers. Nothing will be available at this switchpoint unless there are signals on the tape. Since we have not recorded the first track yet we'll put off a formal discussion of this entire line until we cover the controls in the Overdub (page 40) and Remix (page 49) sections, but we ask you to "track" the line back to the recorder now just for "drill". The large block diagram on the foldout page (do you have it visible now?) shows the entire playback signal pathway from the recorder to this connection, and the line continues a little further to show that this signal also appears at a back panel RCA connector marked TAPE OUT.



Output impedance of this circuit is affected by the internal loads as well as the connection, so a safe value to use in calculations is 1.7 k ohms. If you are not actually using this output, DISCONNECT THE CABLE and the performance of the PORTASTUDIO will improve. Nominal output level is 0.3 volt (−10 dB). Maximum output before clipping is 5.6 volts (+15 dB). Because of this internal wiring, it is not necessary to patch the

TAPE OUT jacks to the mixer MIC/LINE INPUTs. However, special circumstances might warrant the use of a "patch" instead of the much simpler switch selection.

We will have at least three basic jobs to do that will require the playback signal:

1. Simple playback to judge a performance, requiring no corrective EQ. In short, what did you record?
2. Simple playback into a "cueing" system so partially completed tapes can be finished. This function should somehow combine the playback signals with "new" mic signals so musicians may hear a balance of both when "overdubbing".
3. Final remix, when the full control capability of the system (EQ, effects, etc.) can be used to "fine tune" the completed master.

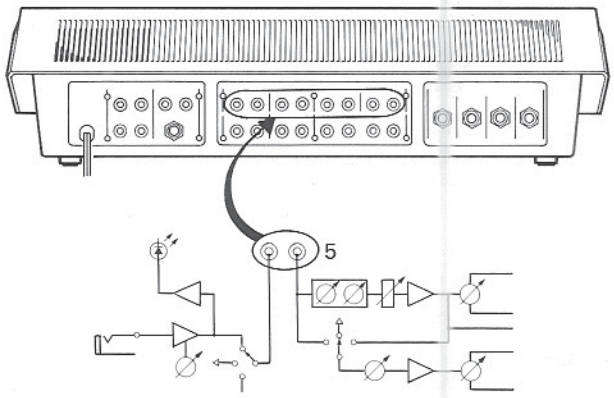
Three basic tasks, one tape playback signal, so, to avoid resetting all the controls on the input module, and "losing" the EQ and record level settings that have taken much time to get "just right" every time you change from record to play you use the Tape Cue System instead of switching the INPUT Selects to TAPE. From this single connection, the playback signal can now be made available to the necessary subsystems.

Because there are separate Cue controls for buss signal and track playback, when you are working on a tape that has 3 tracks recorded, you can still use all 4 mics to record the last track. On the PORTASTUDIO, "tape playback" does not progressively "eat up" your mixer flexibility.

Selecting TAPE on any or all input modules will not disable the functions of the Tape Cue circuit or the TAPE OUT jack. Tape Playback signal will be available on all circuits at the same time. There are several advantages that this feed offers that are added to the normal functions.

1. In remix, the Cue function is not needed, so its stereo "mix" can be used instead as an extra effects send.
2. During the course of normal multitrack production, a good "take" may be acceptable in every way except one; some doubt may arise as to the "mixability" of one track. Since the Tape function may be selected one input module at a time, a single track may be played back, routed through the EQ section and a correction tried out to make sure that re-recording is not required. This checkout will only require the readjustment of one module. All of these concepts involve the use of more controls than we have covered so far, so we'll return to the wiring order and go on.

5. ACCESSORY-SEND-RCV RCA Jacks



The high gain provided by the mic preamplifier allows us to place a "patch point" in this useful location. Since there is an individual "patch" on each module, you can use an accessory device on one line and not affect the other three. The level at this "send" is −10 dB, 0.3 volt, nominal. The minimum load impedance is 1.7k ohms. This output and its performance are affected by loads connected to the TAPE OUT jack, so calculate your total loads with care if you must use both at the same time. If you don't know how to calculate a "load" or even what load we are talking about, read the section on Impedance Matching that begins on page 10. It will tell you what a "load" is and when and how to deal wiht this calculation problem if you are unsure. A limiter connected to this point in the PORTASTUDIO can be set to a range of compression that will not be altered when the fader is moved, or the EQ is adjusted. (Cable route is: ACCESS-SEND to Limiter In. Limiter Out back to PORTASTUDIO ACCESS-RCV)

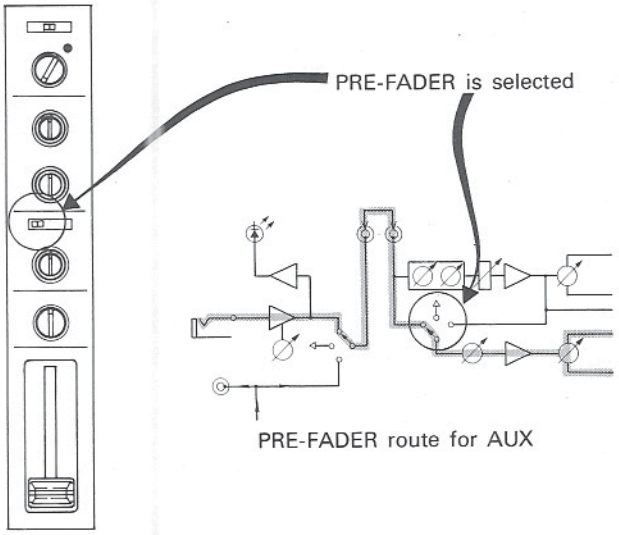
When no accessory device is bridged from "send" to "receive", the jumpers provided must be in place for signal to flow to the EQ amps and on through the PORTASTUDIO. There is no "normal" or automatic internal connection when the jumpers are removed.

Since all the mixing controls lie after the receive jack, it is possible to consider using ACCESS-RCV as a possible line level input, and bypass the switch entirely. This unorthodox "patch" is suggested for use when you are using the special 4 channel record feature to copy a tape from another 4 track recorder, and more time for patching is available. Any successful recording is not likely to need the TRIM or OVERLOAD LED indicator provided on the first amp, and it is also possible that the source recorder will use RCA cables. Using this RCV jack as a "line in" will save you the cost of 4 phone adapters. By-

passing parts whenever possible improves signal quality, and when you are copying you will appreciate all the help you can get. Nominal input level at this "receive" is −10 dB (0.3 V). Max input level will be +15 dB (5.6 V). The input impedance of the RCV jack is 68k ohms.

6. PRE-FADER SELECT for "AUX"

Right after the accessory send-receive patch point, the block diagram shows a line leading to one side of another three way switch. This is the first appearance of the Aux system.



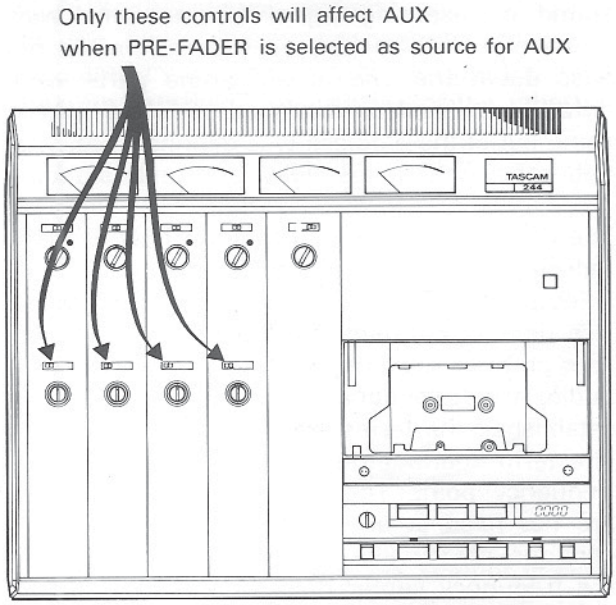
Using this switch setting to feed signal into the Aux system will give you a separate mix that will not be altered when you

1. change the tone with the EQ section, or-
2. alter the level sent to the main line (Buss L + R MASTER Fader).

The block diagram shows that the signal leaves the module main line before the EQ section, goes first to a separate volume control marked AUX-GAIN, then through an amp to the AUX-PAN. After the PAN two lines are shown running up and down the page, and they show that there are 3 other signals (one from each Input Module) which may also appear in this "system". Is there a "Master" for overall control of this mix? No, there are no more "pots" (potentiometers) shown on the block before the AUX-SEND RCA jacks (two), so you will have to adjust this "mix" with the individual controls on the modules. However, there is a line pair shown that does go to another part of the PORTASTUDIO. The lines drop down the page and enter the Buss Monitor group that selects what signal you will hear in the headphones. You will be able to hear the arrangement of signal on this subsystem in the phones.

Two problems in the comprehension of the PORTASTUDIO as a system are now beginning to appear.

1. WHERE ARE THE PARTS ON THE UNIT?
   After all this discussion of the block diagram, you can understand the way that the system is supposed to function, but now, you must find the controls on the top panel. A "trick" illustration will solve this problem. Here is a drawing of the PORTASTUDIO showing the controls that will affect the Pre-Fader Aux System signal on its way out of the unit.



2. WHAT IS THIS LIMITED SYSTEM USED FOR?
   Some examples of use will help here. This pre-fader source can be used to feed a cue system with a signal that will remain at a constant level, regardless of what you do with the Input or MASTER Faders. A pre-fader feed will work even if both faders are completely "off" and can be used to check an input without recording it. In remix, this type of "send" will allow you to have a signal "walk back" into an echo device when you lower the module fader. As the regular signal diminishes the echo send remains constant, and produces this dramatic effect. There are other uses possible for this Aux system if the POST Fader setting of the switch is selected, but we have not yet reached the point on the module where the other options are shown, so let's wait until we get there before completing the Aux System explanation.

7. Two Section Semi-Parametric (Sweep) EQualizer

Before we begin, the label itself will need some

explanation. What is parametric and, equal to what? An obvious first question, because the term does not describe what you do with the controls. In multitrack audio, tone controls are almost always used to "make different" and the concept of "make the same" doesn't quite fit. How did the term "equalizer" come to be used in audio tone control?

The Telephone Company uses it.

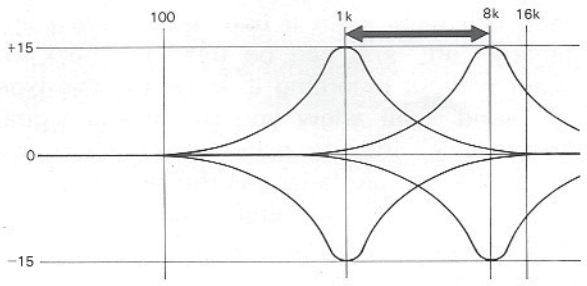
In the early days, the telephone worked well in the lab, short runs of 100 yards or so but....

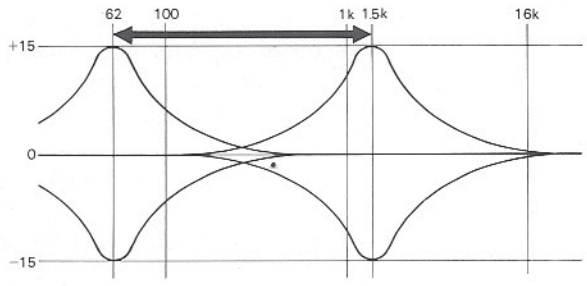
When two "phones" were 10 miles apart, the line between them did not transmit all of the sound representing signal in the same way. Some parts of the frequency spectrum did not pass down the line at all. Some parts were different in level or displaced in "time". What came out of the earpiece was definitely not what had gone in 10 miles away, and understanding a conversation proved to be difficult. What now? The Phone Company had to learn how to make the output "sound like" the input.

If "output equals input" is the concept, an "equalizer" is a logical name for the device used to fix your problem. Just as in many other concepts in audio, the telephone company language has established the terms we use today.

The term "parametric" refers to the adjustable frequency point. The "parameters" or "rules" are not fixed at any specific number, but are continuously variable. Two aspects of the circuit, the frequency center point and the "boost" or "cut" in gain are adjustable without "steps".

Here are two graphs showing the control ranges of each of the two sections. Each section provides some "overlap" of the previous section in regards to the frequency range.





The great advantage of a parametric or continuously "tunable" equalizer over the more conventional "fixed center frequency" types is that you can adjust the frequency center point to the precise area you need and then the cut or boost you use will be more effective. You get the result needed with less rotation of the control, and this puts less "strain" on the electronics. No matter how many "frequencies" there are on a "fixed number" type EQ, it is unlikely that any one will prove to be "just right" and many more ranges are needed to cover the bandwidth needed to do the job. More sections means more parts to degrade the signal, and more money to build the EQ. In the long run a "parametric" EQ requires fewer parts to cover the same task, so it costs less, and in many cases will work better than large multisection graphic EQs. These devices are capable of very sophisticated tonal modification, but may leave many sections unused on the average job. To achieve a comparable "spec" a graphic has to use superior parts to counteract the effect of all those extra amplifiers. In audio processing, "less" is always the better working concept, so use the brute force EQ after all other methods have been explored, move the mic, change the mic, and finally — try the "cut" functions of the EQ first.

Even experienced engineers have a tendency to forget that "cutting" the lows will have a similar effect to "boosting" the highs, and puts less of a strain on the electronics. The results are not identical, but they are close enough to warant trying. Cut bass, raise the overall gain and see if it sounds better than just "boosting" the highs. On the block diagram and gain chart you can see that the EQ stage has a moderate "boost and cut" range (±15 dB) and a substantial reserve, or headroom (10 dB). The reserve in the circuit is necessary to maintain sufficient "headroom" when the EQ is set for maximum effect. Without some extra margin, you would have to lower the TRIM to avoid overloading subsequent sections of the PORTASTUDIO when using the extreme boost settings of the EQ.

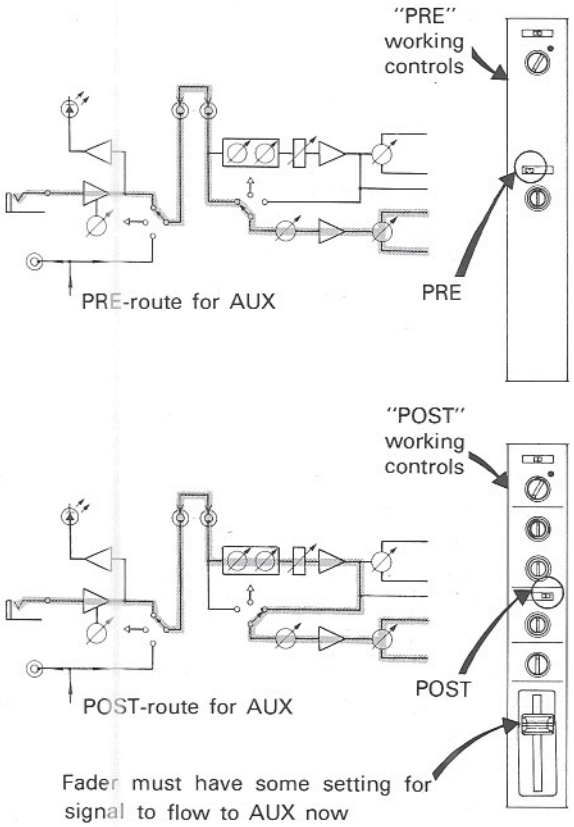
8. INPUT Fader

The main mixing control for individual signals on the PORTASTUDIO, Faders, also called "pots" or attenuators always cause loss in order to control signal. Gain stages in an electronic device always run "wide open" at whatever gain they are set for, unless they have provisions for TRIM or actual gain adjustment. In the PORTASTUDIO, only one of the many amplifiers used actually has TRIM, the first stage in the input

module. When you advance any straight line fader on the PORTASTUDIO you are just reducing the loss it causes. The entire signal flows to the next stage only if the fader is "wide open" or up all the way.
After the signal has passed this section, the block diagram shows two possible pathways for our signal that may be used individually, or both at the same time. Here is the line that allows the POST Fader and POST EQ option for the Aux System so we'll return and finish the explanation of the AUX controls.

9. AUX System Select Switch

We have made mention of this AUX system when we first encountered a possible pathway. Now, with both "options" in view, a review.

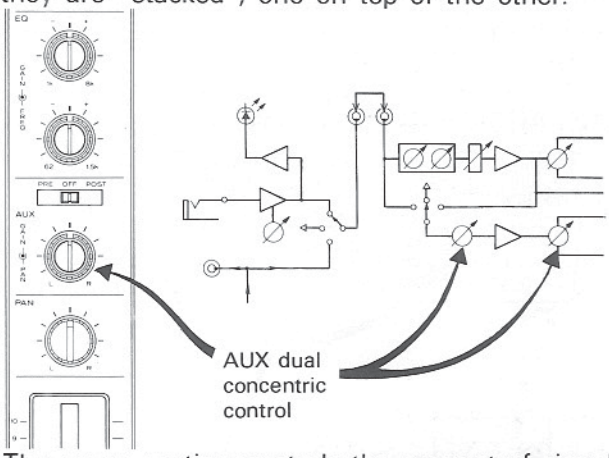


AUX System Select Switch has three positions:

- Set Left to PRE: Selects a source of signal that will not be affected by the setting of the Input Fader and does not have the result of the EQ settings.
- Set Center to OFF: Mutes the subsystem. The action can be used in a manner similar to the use of the module "mute" already discussed, but in reference to this AUX only. (drop in a "preset, clear all signals without losing their settings, so you can "fine tune" one, etc.).
- Set Right to POST: Selects a source of signal that will have the results of any EQ section settings and that will also "track" the setting of the INPUT fader.
  POST fader is usually used to feed signal to an echo device when a fadeout is planned. As the INPUT fader is lowered, the POST signal will also "fade" and you won't be left with a "ghost player" in your mix. Now for the rest of the "parts" in the AUX line.

10. AUX System Dual Concentric Controls
A. AUX-GAIN (upper section)
B. AUX-PAN (lower section)

On the block, these two controls are shown separated by an amplifier, but on the top panel they are "stacked", one on top of the other.



The upper section controls the amount of signal sent from each module into this system. To increase, rotate to the Right.
The lower section of this control is a "pan". Full left rotation sends signal to the LEFT AUX-SEND (output) jack only.
Full right rotation sends signal to the RIGHT AUX-SEND (output) jack only. Any setting in between will divide the signal proportionally between the two outputs. The stereo effect can be auditioned in the headphone monitor circuit by selecting AUX. We'll show you where that is when we get to the MONITOR (headphone) select switch.

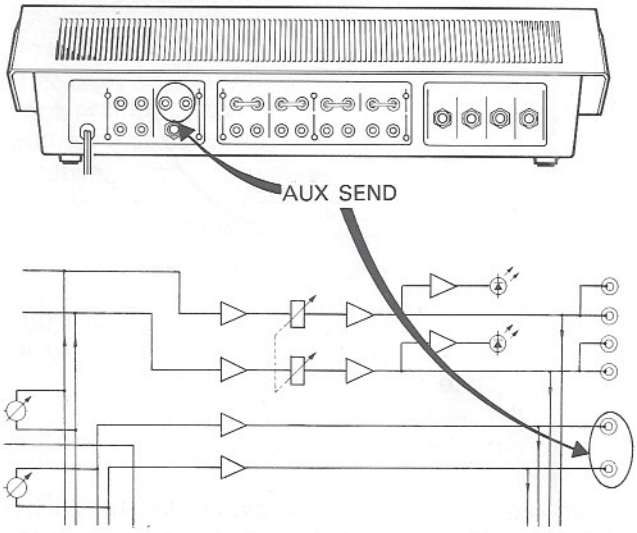
11. Aux Send Summing Amps

These amplifiers don't show on the outside, but their contribution to the system is considerable. These devices allow the four input modules to add their signals together without one module distorting the output of another.
When you wish to "combine" or "sum" two or more varying voltages that are being used to represent sounds, a simple "joining together" of the wires will not work. There is no "one way sign" on a piece of wire. First, you must cause the energy to flow in the direction that you want.

A network of resistors can be arranged in such a way as to make the "forward" pathway the easiest "way out" of the network and the route back in to the adjacent module very "difficult". Simple, but this network reduces the signal level a lot, so this unavoidable loss must be made up by another amplifying stage. This method of resistive net and gain make-up is referred to by engineers as the "brute force" method and is widely used. With no user controls on the outside, you might think that a discussion of "summing networks" would be unnecessary, but understanding these principles will make clear why you can't use a simple "Y" adaptor to add two signals together.

After the summing amps, there are two lines on the block that route signal "down" to the MONITOR (headphone) signal select switch. We'll discuss the options there later.

12. AUX-SEND RCA Jacks



The nominal level at this output is −10 dB, 0.3 volt, maximum level before clipping is +15 dB, 5.6 volt. The minimum output load impedance (lowest permissable number in ohms) is 1.7k ohms.

If we are now ready to send signals out to some sort of special effects device (echo, phaser, flanger, etc.) where does the processed result return to the "mix" so it can be recorded? There are two possible methods, one complex and another that is more straightfoward.

COMPLEX AUX RETURN METHOD

Use the input modules that don't have mics plugged in. much of multitrack production is done on a "one mic per track" basis and will leave you with three unused input modules. You can take advantage of the functions that they provide to do things to the return signal. Separate EQ can be used to improve the effect. The two AUX Outputs can be used to feed one device per jack and the "spare" input modules used to control the selection and balance of the two separately. This method is not restricted to the "one mic per track" jobs; even stereo is possible. Remember that you have a total of four modules. Two for the stereo main signal, two for the effects return and you are "in business".

A CAUTION: Both of these complex patches can lead to a circular assignment, or feedback loop. To use these setups successfully, the AUX GAIN on the Return Modules must not be used. The simplest way to make sure that you don't assign the processed signal return BACK OUT TO THE EFFECTS DEVICE by accident is to be certain that the PRE-POST switch for the Aux signal select function is set to the center, or OFF position on the "return" input modules. If the switch is OFF, the AUX GAIN will not work.

A WORD OR TWO OF REALITY

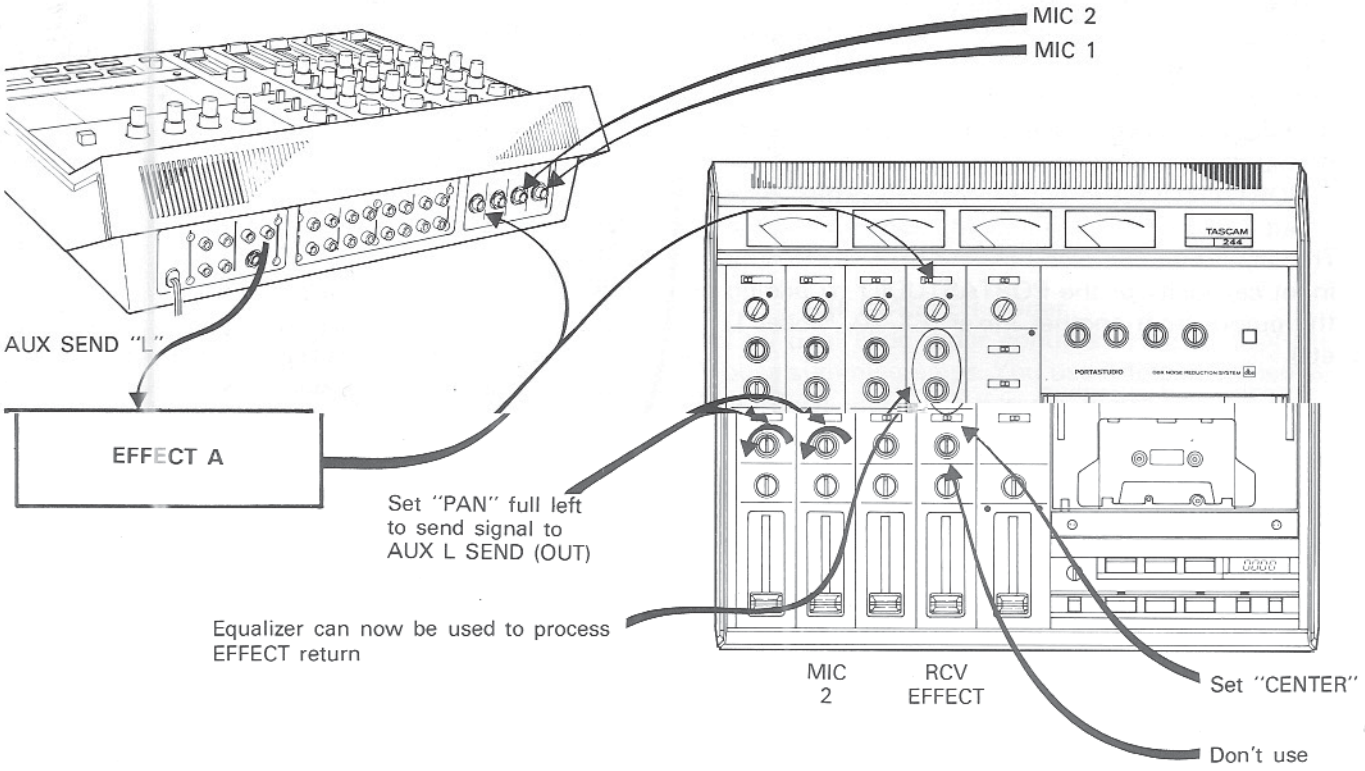
The first time user may say at this point that these extras are so hard to grasp that the benefits are not worth the risks. The PORTASTUDIO is new, there are what seems like a thousand knobs, and the first few pages in the manual at first seem to be describing logics that are so sophisticated that they only make sense to a 20 year "pro". We agree! These mix patches are complex and their routings are not easy to visualize. We will not insult your intelligence by saying otherwise. Take heart. The PORTASTUDIO is a tool, not a toy. Like any good tool, good results depend on practice and understanding. We have provided many sophisticated features But—take your time — it is not necessary to use all of them on the first track, and perhaps your first experiments should be restricted to the simplest routes. You will find a use for this "deluxe patch" when your art is in need of the control that it can provide.

USING AUX IN

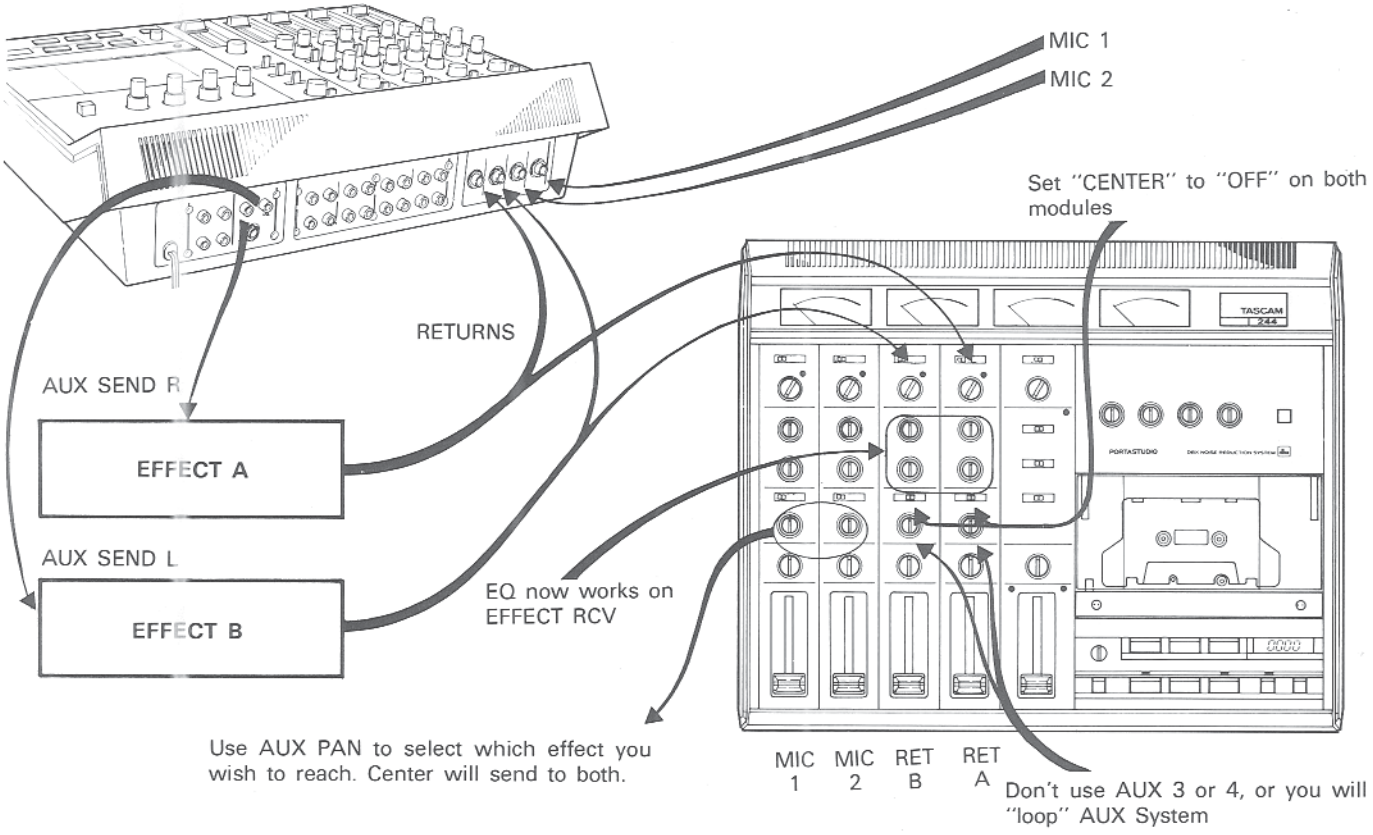
13. AUX RCV IN RCA Jacks
14. AUX RCV Dual Control

The input impedance of each receive RCA jack is 37k ohms. The nominal (expected) level is −10 dB, 0.3 volt. Max input before clipping is +15 dB, 5.6 volt. At full right rotation, the maximum permissable input level is reduced to +9 dB, 2.8 volt. The rules for using this Dual Input circuit are more straightforward. The signal applied to the left input jack goes only to the left Buss Summing Amp, the right jack feeds only the Right Buss Summing Amp. Because

● Single effect using spare input return method



● Double EFFECT (or 1 Stereo)

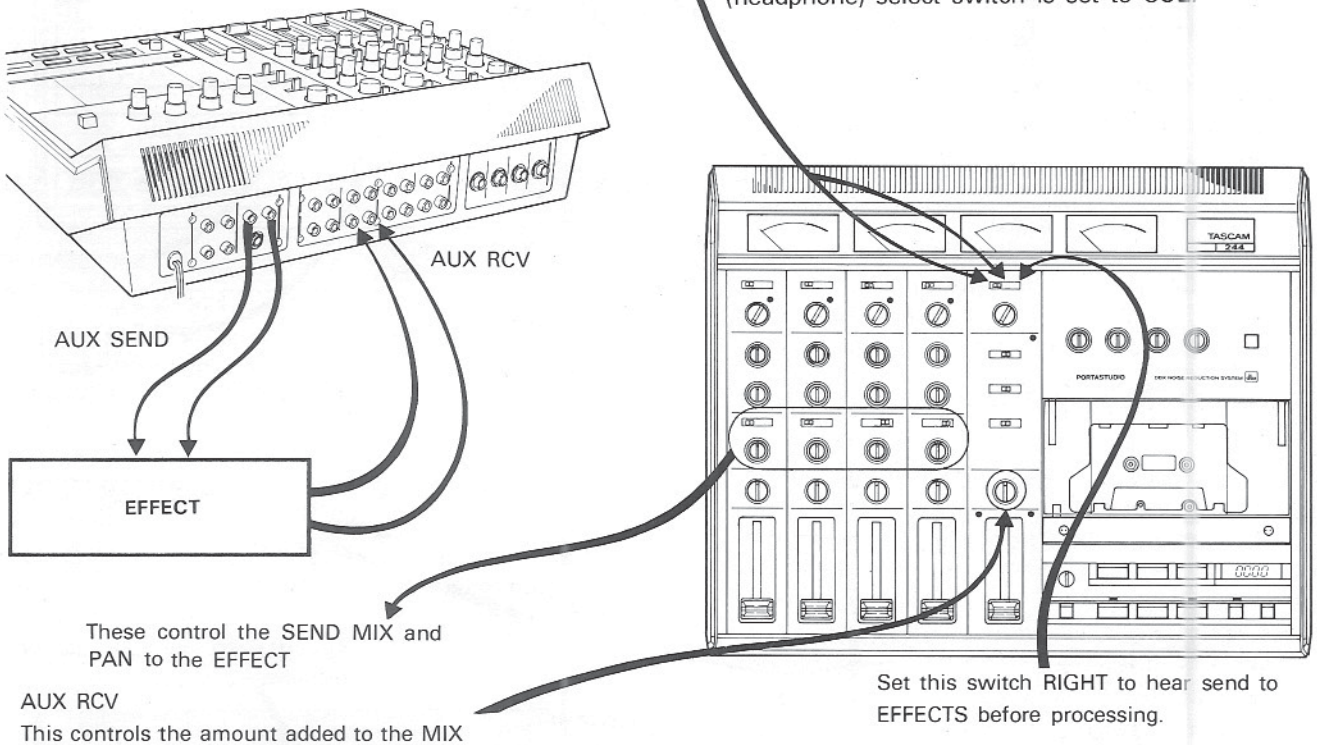


AUX RCV control has two sections, but only one "knob", there is no separate adjustment. If you need to "balance" left side "receive" against right side "receive", there are two things you can do:

A. Does your device have controls on its output?
B. If not, you can use the AUX PAN on the PORTASTUDIO to get a balance on the "way out".

This input pair can also be used to "expand" the input capability of the PORTASTUDIO by adding the output from another mixer (EX-20, Model 1, etc).

When set LEFT (REMIX) you will hear L+R Buss MASTER plus AUX RECEIVE in stereo.
When set CENTER (CUE) you will hear L+R Buss plus AUX RECEIVE and TAPE Cue System as a center monophonic feed.
All the circuits that have PAN pots will work in stereo, but this setting of the Cue system will not allow you to hear the effect. To judge the actual stereo placement you will have to set the PANs while monitoring either REMIX (input PAN) or AUX (AUX system PANs). There is no way to monitor the Cue system PAN function in the headphones. These PANs affect only the TAPE Cue L+R stereo output RCA Jacks on the rear panel, and the internal route to the headphone amps is a mono feed when this switch is set to CUE. The RCA jacks remain stereo when the MONITOR (headphone) select switch is set to CUE.


AUX RCV
AUX SEND
EFFECT
These control the SEND MIX and PAN to the EFFECT
AUX RCV
This controls the amount added to the MIX of whatever your EFFECT has done to the signal you sent out.
Set this switch RIGHT to hear send to EFFECTS before processing.

Before going on to the L + R Master and the Record Matrix Logic, we must return to the Input Module and explain the remaining controls and internally wired "options". There are three items left: and amp, a special signal route, and a control.

15. Input Module Buffer Amp

This amplifier has a gain of 8 dB, but its primary purpose is not really signal "boosting". It is here to isolate the fader from the panpot and to provide the extra current necessary to feed the three-way multiple connection shown on the Block Diagram.

A. Signal feeds the POST position of the AUX Select switch.

B. Signal feeds the Four Channel Record special function and the meter 4 Channel Record Logic. When the Four Channel Record function is selected, each Input Module is assigned to its corresponding track on the recorder, and the four meters now read the signal derived from this point in the wiring module one goes to meter one, and so forth. Although the PAN pots still feed the L + R MASTER fader, it is no longer connected to the recorder. You can still hear "stereo" in the monitor, and the level will be affected by the position of the L – R MASTER, but to the recorder no "mixing" is possible. You will now have one module feeding each track, and the recorder will ALWAYS ERASE ALL FOUR TRACKS PRIOR TO RECORDING ALL FOUR TRACKS SIMULTANEOUSLY! YOU CANNOT MAKE AN OVERDUB IN THIS MODE. The Tape Cue cir-

cuit is disabled, depriving you of the necessary headphone feed, and anything that is already on the tape will be erased when you start 4 Chan Rec, so take care. This special function is provided to allow copies to be made, and to record all four tracks "from scratch". It is definitely not the "normal" method of use. True, it is possible to begin a tape in this mode, you can record one, two or three tracks at one "pass" with this logic, switch to Sync (That is the overdub mode) and finish the remaining track or tracks but if you forget to switch from 4 CH to SYNC, YOU WILL ERASE YOUR FIRST "PASS".

C. The final signal feed from the buffer amp is to the Input Module PAN pot.

16. Input Module PAN Control

This single knob works two faders that are wired "back to back". As you rotate, one fader is turned up as the other is turned down. At dead center, each control is down just enough to compensate for the fact that two sources will have the same signal. This is primarily an active stereo pan requirement. The compensation allows you to "pan" a signal from left to right without getting an unwanted "rise" in sound power as you pass through "center".

In addition to its use as a "stereo" perspective control, this "pot" is used as part of the PORTA-STUDIO signal assignment system. When using the CUE position of the headphone monitor system, the action of the PAN will affect which buss will receive signal, but you won't hear any left to right "motion" when you turn the PAN. The center setting of the MONITOR (headphone) select switch (CUE) is mono. If you PAN to L BUSS and set R BUSS to record, no signal will reach the tape!

When 2 signals are to be recorded independently, make sure that the PAN settings for each input used are "full left or full right" or you will have a "mix" of both the signals on each track, instead of the separate recording you wanted.

To avoid these errors;

1. Check the meters to make sure that the active track (meter light on) is receiving all the signal. (No indication on the "wrong" (unlit) meter indicates that the PAN is rotated all the way L or R and is, in fact, feeding the correct BUSS).
2. When you are recording in stereo, both meters will be lit, so switch the MONITOR (headphone) select switch briefly to REMIX before recording. This action will give you a chance to audition the stereo perspective and confirm that the signal has been assigned properly. We'll go over this again when we review the RECORD FUNCTION switches.

17. L + R Buss Summing Amps

These two amplifiers make up the losses caused by the combining network that accepts signal from six possible sources.

A. The four Input Modules.
B. The Aux Rcv circuit.

18. L+R MASTER Fader

This dual slide fader controls the output of the buss summing amps. You use it to simultaneously adjust these 5 different "feeds"

A. The level shown on the meters.
B. The level sent to the record switching matrix that determines which tracks will record.
C. The level sent to the LINE OUT.
D. and AUX OUT RCA jacks.
E. The level sent to the MONITOR signal select switch when the switch is in CUE or REMIX position.

19. L+R MASTER Fader Isolation Amp

All these places that the signal must go to represent a lot of "load" so another amplifier is provided to isolate the fader from this five way "Y' connection.

20. L+R MASTER LED Amps and LED OVERLOAD Indicators

Several modes of operation will switch the meters away from the L+R MASTER line. These LEDs will alert you to an overload condition on these busses (L+R 5 possible feeds listed above) when the meter logic has been "set" elsewhere in the signal chain.

An example may help clarify the need for these extra LEDs. When you are ready to remix, and you switch the recorder to SAFE by centering all the Record Matrix switches, the meters will read the play back level directly form the recorders first playback amp on a "track by track" basis. (Meter 1 tied to track 1 and so forth). This location as a source of meter signal will not show level changes when the input faders are moved, EQ settings are changed, or the L + R MASTER is moved. To get the meters to read the L + R outputs, you must set at least one track to record-ready, and run the risk of accidental erasure. If this is unacceptable, the LEDs will warn you is your "mix" is getting "too hot".

There are several "logics" for the meter circuit. You will find them listed completely in the Record Matrix section on page 32.

The LEDs are set to light at 0 dB(1 V).

CORRECT METHOD OF LEVEL SETTING FOR THE PORTASTUDIO

Many electronic devices have only a single control for setting an output power, or a listening level. in short, there is only one "right way" to turn up the effect, whatever that effect might be. The PORTASTUDIO is not so simple, and its use as a tool for the manipulation of sound requires the adjustment of many intermediate stages of control. If you have a choice, which knob or slide should you use? What do you set first?

The overall performance of the system will improve if you work your way backwards from the L+R MASTER to the TRIM. There is an optimum sequence.

Since this is our first basic set of instructions, we'll start from the beginning, and "power up" as well.

1. With POWER switch OFF, connect to source of AC power*.
2. Connect a mic to INPUT "4".
3. Make sure that the jumpers are in place across the ACCESS-RCV RCA Jacks on INPUT #4.
4. Set #4 INPUT select switch to MIC/LINE.
5. Set all other INPUT selects to OFF (safe).
6. Set #4 TRIM control to minimum gain by rotating fully leftwards.
7. Rotate the #4 Input PAN control fully to the left (the PAN immediately above the INPUT fader).
8. Set all slide faders to zero.
9. Plug in stereo headphones, but DON'T PUT THEM ON!
10. Now you can turn the PORTASTUDIO ON by depressing the POWER switch.

It is normal for the PORTASTUDIO meters to jump when AC power is first applied, and the headphone amplifier may produce a substantial transient "pop" even if the BUSS MONITOR pot is rotated fully leftwards (OFF). We suggest that you use this conservative approach. Patching in mics and accessories with the power off will insure that you don't damage your ears, the PORTASTUDIO, or any other equipment that you may be using. Take care.

Now lets prepare the monitor circuit so we'll be able to hear what we are doing when we do put on the headphones. To set levels, we'll also have to get the VU Meter Logic set to read the mixer L+R BUSS.

These switches and the explanation of their function have not yet been fully detailed, but we'll get to them right after we set levels and get some experience with the mixer.

11. Set the MONITOR select switch to CUE (MONO).
12. Set the L BUSS select switch to TRACK 1.
13. MAKE SURE THAT THE RECORD FUNCTION SWITCH IS SET FULLY TO THE LEFT (SYNC).

To make this preliminary walk-through more useful, you should have some source of sound. Rather then playing, singing or speaking yourself, we offer this idea: Use a portable radio as a source of sound to experiment with. Select a station that carries the type of material that you prefer, and you won't have to combine the problems of performance with those of learning about this unit. If you work in the audio-visual area, find an all news or talk format station, and you will be able to experiment with speech recording as well. We grant that the fidelity of these recordings may be misleading, but a comparison between the broadcast "as heard" and what you get when you play back will still allow you to experiment with the EQ and the mic positioning. When you are more familiar with the basic functions of the PORTASTUDIO, you can switch to "real sound" with more confidence.

A minor point to remember: Small radios may not have great fidelity, but they do have the benefit of using material produced by very good engineers. Live music may not be as easy to record as the radio.

LEVEL SETTING

14. Set the L+R MASTER slightly below the "marks" (7-6).
15. Set the #4 Input fader slightly below the marks as well.
16. FINALLY—Raise the level of the incoming mic signal by rotating the TRIM control until you get an acceptable reading on the VU meter(s). What's an "acceptable" reading? We'll discuss that when we get to the section on the built-in DBX II Noise Reduction System (page 56). At this point, you should see the meter move, but you won't hear anything, so lets fix that.
17. Put on the headphones.
18 Rotate the BUSS MONITOR control rightwards until you have a comfortable listening level.

*NOTE: See page 52 for voltage conversion, and note for U.K. customers.

This is the first of the many times that you will see that each requirement of multichannel recording requires independent control. In this instance the same signal is being adjusted for two different purposes.

A. An electrical circuit to a recorder is being adjusted to the "correct" level by reading a meter.

B. A listening level is then independently set with the BUSS MONITOR control. Rotating this control does nothing to the meter reading, and recording could take place successfully even if you could hear nothing. A good grasp of this concept is the key to succesful multichannel engineering.

Now the explanation of the sequence.

The idea in setting the TRIM control is to use the smallest amount of "gain" from the first stage that will still give an acceptable reading on the meter. Less clockwise rotation of the TRIM means less gain is set, and you will have more headroom or reserve power left in this stage. It will produce less noise as well.

A word about "noise": The specifications for the mixer section of the PORTASTUDIO are given assuming an input level (MIC) of −60 dB (1 mV.), then the slide faders are set to their "marks" and the "noise" is measured. Any signal that is more powerful than −60 dB will allow you to lower the gain. (start with the TRIM, it contributes the most noise).

21. L+R LINE OUT
22. L+R AUX OUT

Although this is the final stage of the mixer section , we don't recommend using it as a listening point while you are working on a tape. The reason? If you look at the block diagram, you will see that if you use this circuit for "monitor", you can't audition the Aux circuit. The Cue circuit does not normally feed in to this output either, so to play back a track in order to make an overdub, you will have to switch an Input Module to TAPE and reduce your mixer flexibility in several ways. In addition to these difficulties, you will see that there is no separate L+R MASTER for these jacks. Whatever level you must set on the meter to get a "good" recording will be the level you send to your "outside" amplifier. If this amp does not have its own volume control, you will not have any way to adjust the listening level except to use the L+R MASTER and that will compromise the quality of your recordings.

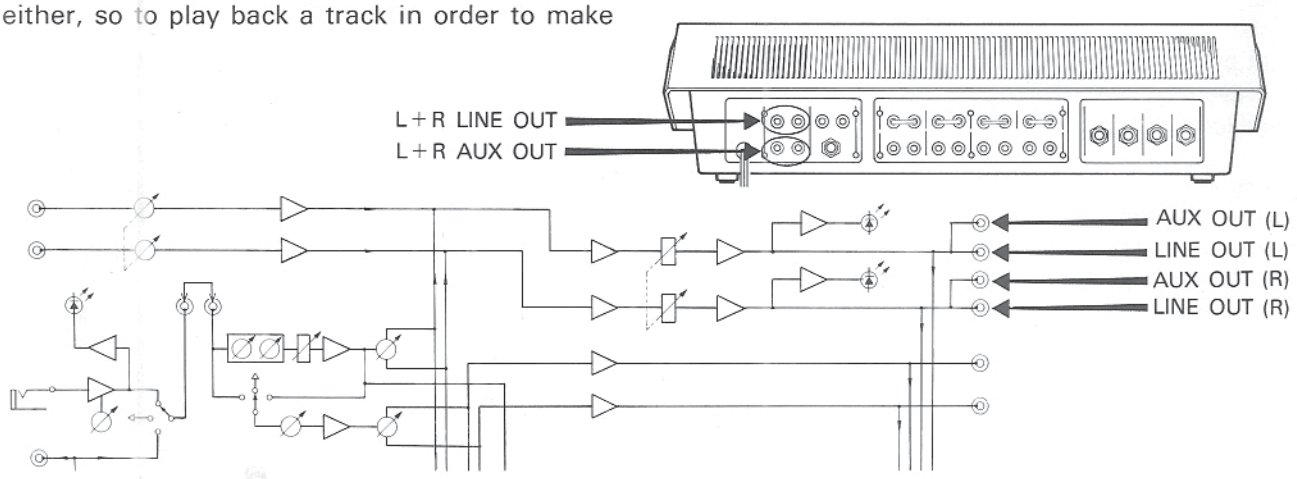
The use of a receiver or integrated amplifier that has a separate level sources for signal ("Aux" or "Tuner" will probably be compatible) will allow you to use loudspeakers as monitor for your final mic, but the level control for the loudspeaker system will still be on the receiver, not on the PORTASTUDIO. The L+R MASTER will now be needed to "feed" your "mixdown" recorder. If you plan on using loudspeakers as a part of your working method, make sure your integrated amplifier is within reach when you "lay out" your system.

We suggest that when you are working with a headphone driven and controlled by the Monitor circuit in the PORTASTUDIO, that you disconnect the LINE OUT and AUX OUT until you need to use them. Just turning off the units or turning down their controls may not completely remove them from the circuit, so the "purist" approach is to unplug the cable at the PORTASTUDIO end. This will eliminate the possibility of having the disconnected cable become an antenna and pulling the local radio stations into your mix.

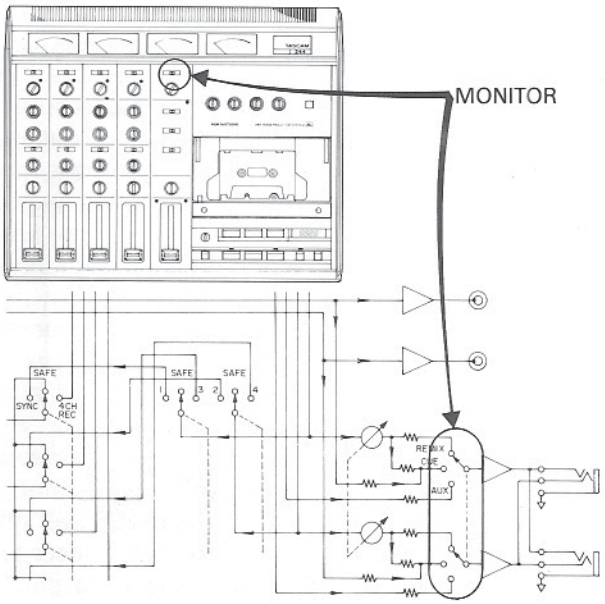
Output impedance of these curcuits is 100 ohms. Nominal output level is −10 dB, 0.3 volt. Max output before clipping is +15 dB, 5.6 volts. These signals appear on the following meters only:

L LINE OUT and AUX OUT = Meter 1

R LINE OUT and AUX OUT = Meter 2



23. MONITOR (Headphone) Switch and its Logic



What signals you will hear in the headphone circuit will be controlled by the setting of this switch.

- Set Left — REMIX: This setting selects only the L+R Buss Output from the mixer and feeds one buss to each "side", this is a stereo feed. This setting is provided to allow independent use of the TAPE CUE controls when you are mixing down a completely recorded tape, and you don't need to combine the new signals from the mixer with the old signals already on the tape. NO TAPE CUE SIGNAL IS HEARD IN THIS MODE, unless you route it to a device (Echo, whatever) and then return the processed signal to the mixer via AUX RCV or an Input Module.
- Set Center — CUE: This setting combines the L+R Buss Output with the Tape Cue signals to form a Center Feed Mono monitor that works this way. The signal you will hear will be a mono combination of the L + R BUSS mix and the sum of the 4 TAPE CUE signals set by the upper section of the dual concentric controls in the TAPE CUE row. These knobs are located directly above the transport. There is a separate section provided for each track in the recorder. TAPE CUE controls will only have signal available after something has been recorded, and will not "set" levels when you assign a BUSS to a track on the recorder. They are playback only, and the PAN function cannot be auditioned in the headphone circuit. PAN effect appears only at the RCA L + R jacks on the rear panel, and is provided for stereo use of this circuit when it is not needed as an overdub guide to what is on the tape. Unless you have a device attached to these RCA jacks, the PAN settings may be ignored.

A CAUTION: The PAN settings on the INPUT Modules determine which BUSS the module is assigned to L or R, and this Mono Cue will not warn you when a mis-assignment has been made. Here's what can happen.

When using the CUE position of the headphone monitor system the action of the INPUT PAN will affect which buss will receive signal, but you won't hear any left to right "motion" when you turn the PAN. If you forget, and leave the INPUT PAN set left to L BUSS when you want R BUSS to record, you will still hear a signal in the monophonic headphones but— no signal will reach the R buss and when you play back the "take" there will be nothing recorded on the tape! If the PAN is set part way between the two busses, the error may be even harder to understand.

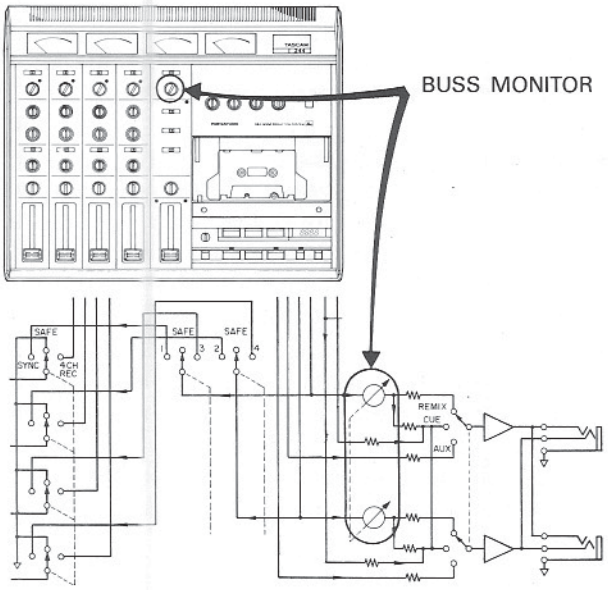
Something will record on the new track, but the actual level on each BUSS will be affected by the PAN pot position. When you play back, the result can be anything from slightly low to almost inaudible! Even if you read the active meter, and make up the loss caused by the mis-set PAN. Your "make-up" gain is not the right idea. You will be increasing the noise and your recording will be compromised. These "checkouts" may help.

1. When recording one track at a time, check to make sure that the active track (meter light on) is receiving all the signal. (No indication on the "wrong" (unlit) meter indicates that the PAN is rotated all the way L or R and is, in fact, feeding the correct BUSS)
2. When 2 non-stereo signals are to be recorded at the same time, make sure that the PAN settings for each INPUT used are "full LEFT or full RIGHT" or you will have a "mix" of both the signals on each track, instead of the separate recording you wanted.
3. When you are recording in stereo, two meters will be lit, so switch the headphone select switch briefly to REMIX before recording. This action will give you a chance to audition the stereo perspective and confirm that the signal has been assigned properly.

- Set Right — AUX: This setting allows you to adjust the stereo mix made up from the AUX-GAIN and AUX-PAN controls on each Input Module before routing it out to an outboard processor. Because of the Aux capability to

"mix" Pre-Fader, and thus avoid sending a signal unavoidably to the PAN and then to a Buss, we can use this circuit to develop our Fancy Cue Systems described in the Overdub section on page 42. Don't miss it.

24. BUSS MONITOR Level Control

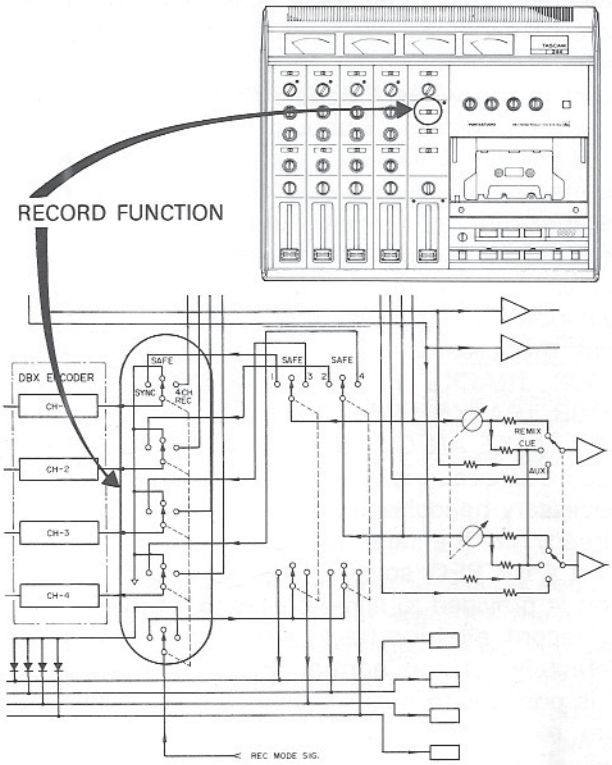


The block diagram shows that the L+R Buss signal is fed to the BUSS MONITOR Volume control directly, and the Tape Cue mix lines go directly to the switch, and then to the headphone amp. What this means is that the Tape Cue signal levels are controlled individually by their own "pots", there is no overall "playback" master. Only the L+R signals are affected by the BUSS MONITOR control, and because it is a "buss" or final mix cue system, you have no balance control. Voice and guitar balance will have to follow the requirements of the recorder or you will compromise your tape quality. In short, don't turn down the guitar to "balance" the voice, leave that job for mixdown. When you are recording one track at a time, most of these problems will not appear, there are ways to solve these balance problems but they will cost you some mixer flexibility. See page 42 in the Overdub section for Fancy Cue Systems.

RECORD FUNCTION SELECT SWITCHES AND THEIR LOGIC

At this point we have run a signal completely through the mixer and all its accessory functions such as EQ, Aux and Monitor. All that it is necessary now is to go over the settings and conditions of the record control circuits so the recorder will "print the signal" on the tape. These three switches interact, and there are lot of "if this is on, that won't work" kind of statements in this section, so take your time reading.

25. RECORD FUNCTION Switch



This uppermost switch has three positions and is used to select the overall mode.

- Set Left to SYNC: One or two tracks Maximum will be possible, with the meter logic and the record-ready status controlled by the two switches immediately blow this one. YOU SHOULD CONSIDER THIS AS THE "NORMAL" SETTING OF THIS CONTROL.
- Set Center to SAFE: Positive Safety, when in this position the PORTASTUDIO will NOT ERASE OR RECORD ANY TRACK no matter what you do with any other control in this matrix, or on the recorder control panel.
  When the lower switches are both "set" to SAFE, the normal meter logic is to read the individual tape playback levels. This logic is "Locked in" when the RECORD FUNCTION switch is set center, to SAFE. The meters will

read playback levels and all four meters will light.

- Set Right selects 4CH REC: The extra warning LED lights to tell you that this is the All Record, All Erase mode that is used to record "copies" or all 4 tracks simultaneously. We now repeat the warning paragraphs that you have seen in text twice before. This mode is useful, but do you really want to use it? Be sure you understand.

FOUR CHANNEL RECORD Special Function and the METER 4 CHANNEL RECORD LOGIC

When the FOUR CHANNEL RECORD function is selected, each Input Module is assigned to its coresponding track on the recorder, and the four meters now read the signal derived from this point in the wiring, Module One goes to Meter One, and so forth. Although the PAN Pots still feed the L+R MASTER Fader, it is no longer connected to the recorder. You can still hear "stereo" in the monitor and the level will be affected by the position of the L+R MASTER, but to the recorder, no "mixing" is possible. You will now have one module feeding each track, and the recorder will ALWAYS ERASE ALL FOUR TRACKS PRIOR TO RECORDING ALL FOUR TRACKS SIMULTANEOUSLY! YOU CANNOT MAKE AN OVERDUB IN THIS MODE. The Tape Cue circuit is disabled, depriving you of the necessary headphone feed, and anything that is already on the tape will be erased when you start 4 CH REC, so take care. This special function is provided to allow copies to be made, and to record all four tracks "from scratch". It is definitely not the "normal" method of use. True, it is possible to begin a tape in this mode. You can record one, two, or three tracks at one "pass" with this logic, switch to SYNC (That is the Overdub mode) and finish the remaining track or tracks but if you forget to switch from 4CH REC to SYNC, YOU WILL ERASE YOUR FIRST "PASS".

26. 4CH REC LED

This warning indicator has two states.

A. Blinking: Indicates that the logic for the meters and the mixer has been re-routed and the recorder section is ready to record all four tracks simultaneously when/if you press REC and ▶ (Play) on the transport controls.

B. On continuously: Indicates that a 4 channel simultaneous record is "in progress".

27. Upper Buss Assign Switch, BUSS L

Since the PORTASTUDIO Mixer has only two main outputs that reach these select switches, only two switches are provided, one for each buss. This switch assigns BUSS L to either TRACK 1 or TRACK 3. It can only select one of these tracks at a time, so you cannot record 1 and 3 simultaneously.

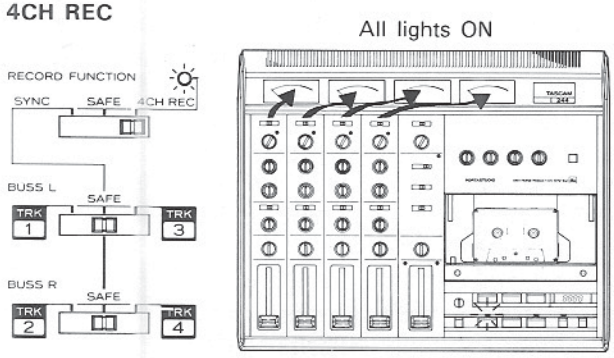
- Set Left: Assigns Buss L to TRACK 1, causes the transport record warning LED to flash, thus indicating "record ready", lights Meter 1 only.
- Set Center: SAFE, disables recording on tracks 1 and 3. If lower switch is also set center, meters will read "playback" on all tracks. DOES NOT MAKE SAFE IF 4CH REC MODE HAS BEEN SELECTED BY UPPERMOST SWITCH IN THIS SECTION! Take care!
- Set Right: Assigns BUSS L to TRACK 3, causes the transport record warning LED to blink, lights Meter 1 only. The meter logic always uses meter number one for the left buss. Meter 3 will only correspond to track 3 in playback when both select switches are set center to SAFE.

28. Lower BUSS Assign Switch, BUSS R

- Set Left: Assigns Buss R to Track 2, causes the transport record warning LED to flash, thus indicating "record ready", lights meter 2 only.
- Set Center: SAFE, disables recording on tracks 2 and 4; if middle (BUSS L) assign switch is also set center, meters will read "Playback" on all tracks. DOES NOT MAKE SAFE IF 4CH REC MODE HAS BEEN SELECTED BY UPPERMOST SWITCH IN THIS SECTION.
- Set Right: Assigns Buss R to Track 4, causes the transport record warning LED to blink, lights Meter 2 only. The meter logic always uses meter number two for the right buss. Meter 4 will only correspond to Track 4 in playback when both select switches are set center to SAFE. Here is a group of illustrations showing the Meter Logic, which meters light and what they read when the various switches are set.

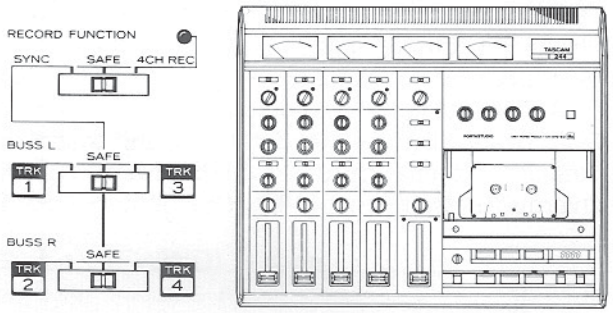
METER LOGIC

4CH REC



Meters read INPUT Modules, not Buss L+R

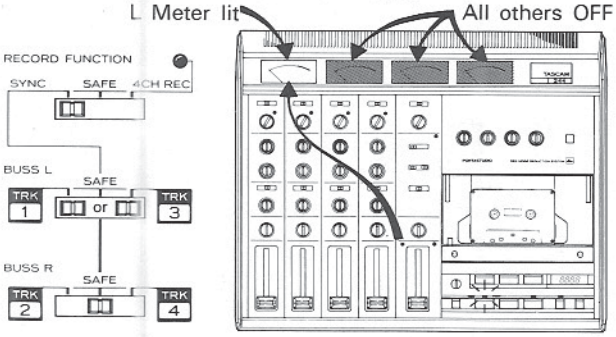
ALL "SAFE"



Meters read Tape Playback Levels

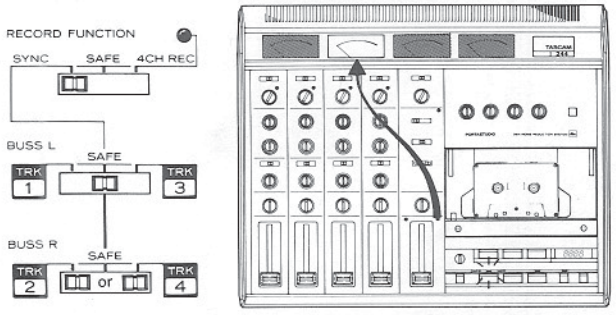
All "SAFE" is determined SOLELY by the top switch. Lower switches may be set to any position and still be safe.

L Buss Set either 1 or 3 REC "Ready"



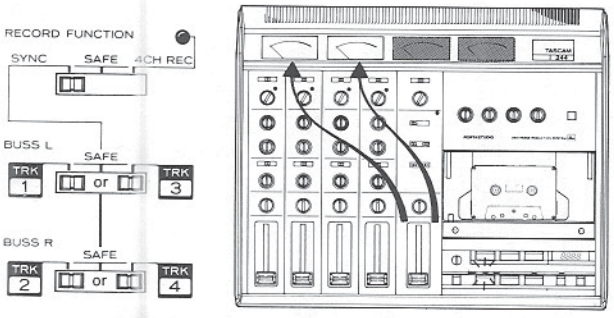
Only one light is ON, but two meters work

R Buss set either 2 or 4 REC "Ready"



Only one light is ON, but both meters 1 and 2 show signal

When both function switches are set to either side, two meters are lit...



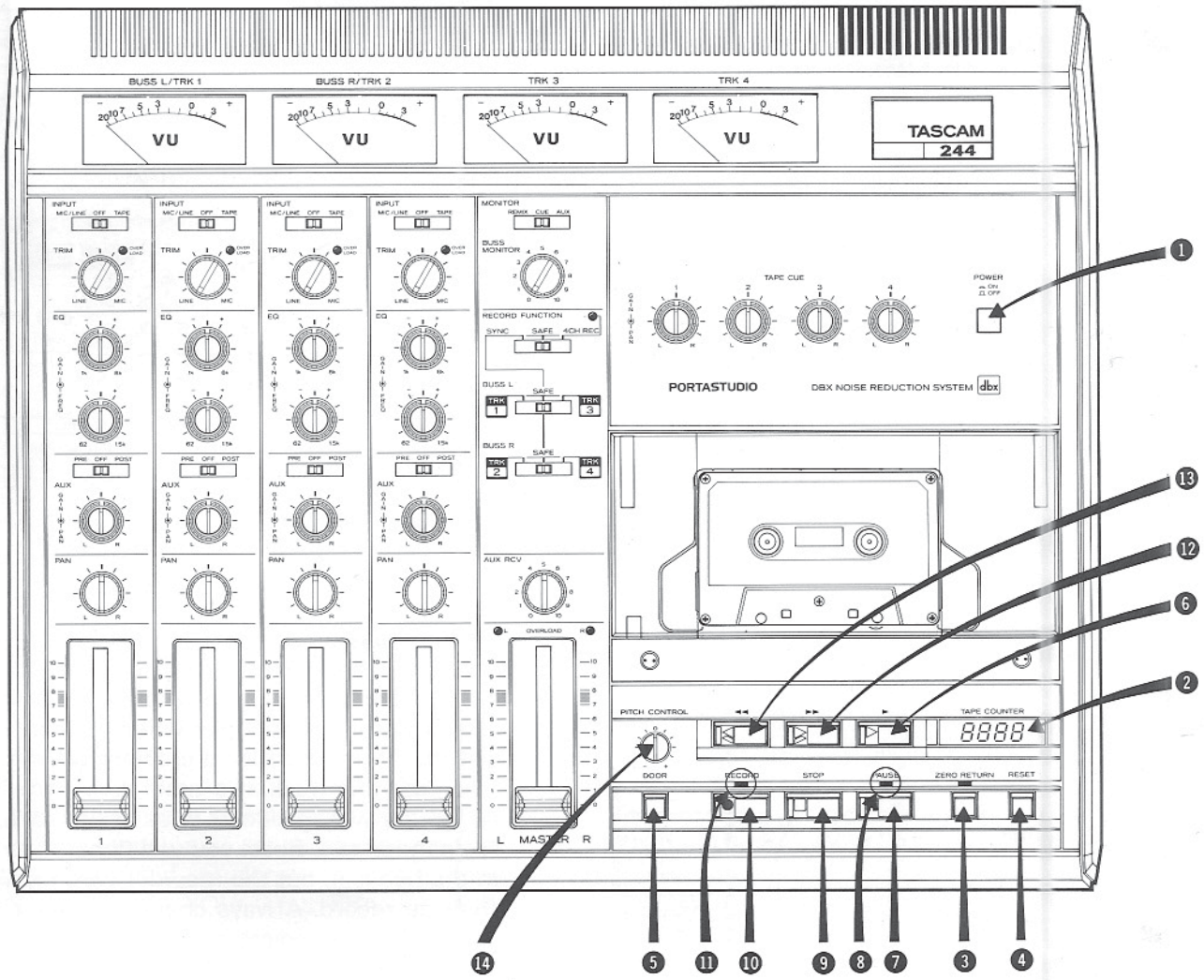
L Buss is Meter 1 for either TRACK 1 or 3
R Buss is Meter 2 for either TRACK 2 or 4

The position of switches in the assign section does not affect the monitor logic for the L+R signals. If signal is present on a buss you will hear it in the headphones when you use REMIX or CUE. This can cause a problem when you are recording one track at a time, if you are "panned" to the wrong side (R Buss assigned, but you are panned to L Buss) you will "hear" the signal in the headset, but the recorder won't have anything to record. Always check the meter that is "lit" for a good indication of level before "punching in" and you will avoid dissapointment.

TAPE TRANSPORT SECTION

We are now ready to explain the controls on the Tape Transport section of the PORTASTUDIO. The only sections of the mixer that remain to go over are the TAPE CUE, the TAPE CUE OUT RCA jacks and the Headphone Tip-Ring-Sleeve Dual Output on the Front Panel of the unit. The details of these circuits will be found in the Overdub section on pg 40. The built-in DBX II circuits show as the last parts on the block diagram, but since they cannot be bypassed, and have no controls on the outside, we will place the information about their action in the "Extras" section that begins on pg 56.

NOTE: Before any serious use of the PORTASTUDIO, we think it wise to recommend that you CLEAN AND DEMAGNETIZE THE TRANSPORT. For details on these "housekeeping" procedures, see pg 52.



❶ AC POWER Switch
Push to turn on, push again to turn off. connect power cord to AC power*.

❷ Four Digit TAPE COUNTER
The use of a four digit counter allows for greater precision in tape location. The Zero Return feature is triggered when 0000 is reached.

❸ ZERO RETURN Button
Push to engage, push again to clear. When this button is down, the tape will stop rewinding at counter 0000. This feature works when rewinding only and will not "seek" 0000 in the fast foward mode.

❹ RESET Button
When depressed, resets counter to 0000.

❺ DOOR Button
When depressed, raises protective cover on the cassette compartment.

*NOTE: See page 52 for voltage conversion, and note for U.K. customers.

❻ ▶ (Play) Button
When depressed alone, starts the transport in Play mode.

❼ PAUSE Button
When depressed, stops' transport from Play or Record but holds the RECORD LOGIC. Press ▶ (Play) again, and recording will resume with a "one button" action.

❽ PAUSE LED
When this LED is lit, a Pause condition is in effect. See the statement in the previous paragraph for the logic.

❾ STOP Button
When depressed, stops transport from all modes and disables all record or pause logics.

❿ RECORD Button
Does nothing when depressed alone; when depressed with
A. PAUSE-REC: Will make transport Record Ready. If top RECORD FUNCTION SWITCH IS SET AWAY FROM CENTER (SAFE) recording will begin when transport enters Play mode.
B. ▶ (Play)-REC: When these two buttons are depressed simultaneously, the transport will enter Rec-Ready and begin rolling tape. If a L + R BUSS RECORD SELECT SWITCH IS SET AWAY FROM CENTER (SAFE) recording will begin immediately on the tracks selected. If a L + R BUSS RECORD SELECT SWITCH IS SET LEFT OR RIGHT WHILE THIS REC-PLAY MODE IS IN FORCE, (RED LED ON OVER RECORD) RECORDING WILL COMMENCE ON THAT TRACK (punch-in).
If a RECORD FUNCTION SELECT SWITCH IS MOVED TO CENTER WHILE THIS REC-PLAY MODE IS IN FORCE, THAT TRACK WILL STOP RECORDING BUT THE TRANSPORT WILL CONTINUE IN REC MODE (punch-out).

WARNING FOR 4CH: *If the 4Ch Record feature is active, it will override the Safes on the L+R BUSS Record switches and Play-Rec will enter Record on all 4 tracks at once. Check to make sure that the 4CH REC LED is off before you start a Record operation on that uses fewer tracks or you may erase important material. (see pg 47 for details on 4Ch)*

⓫ RECORD Indicating LED
This LED has three modes:
A. Out: SAFE, recording will not take place.
B. Blinking: Record ready on one or more tracks, or all record in 4Ch mode.
C. On, Steadily: A recording is in progress, or an active Rec-Pause condition is in force, and you can commence or continue recording by pressing the ▶ (Play) button alone.

⓬ ▶▶ (FAST FOWARD) Button
High speed foward tape travel. Does not respond to the counter Zero Return feature.

⓭ ◀◀ (REWIND) Button
High speed tape travel, counter stop feature is detected in this mode. If set, tape will stop at approximately counter 0000.

⓮ PITCH CONTROL
This unique feature operates in *Record* as well as *Playback*, allowing great flexibility in creative work. the maximum deviation from the PORTA-STUDIO basic speed of 3-3/4 IPS is approximately 15 % fast or slow. In inches per second, this translates into approximately 3-3/16 IPS when the control is rotated fully leftwards (counterclockwise) and approximately 4-5/16 IPS when rotated fully rightwards (clockwise). A center detent locates the "standard" 3-3/4 IPS speed.
Because of the small size of this control, the circuit may prove somewhat tricky to adjust. "Ease of tuning" has been compromised in order to provide the greatest possible range of speeds. We suggest that "full slow or full fast" be used only in final playback, as a minor drift in this circuit from hour to hour may result in a minor change in speed. If you use "maximum" you will not be able to make a minor correction because you will have no "slack." When tuning it is also advisable to allow a "run" of several seconds in play mode for the speed to stabilize, expecially when the change in speed is large. Before beginning to record again check the intonation carefully with a short "play" and you will have less trouble with "drift."

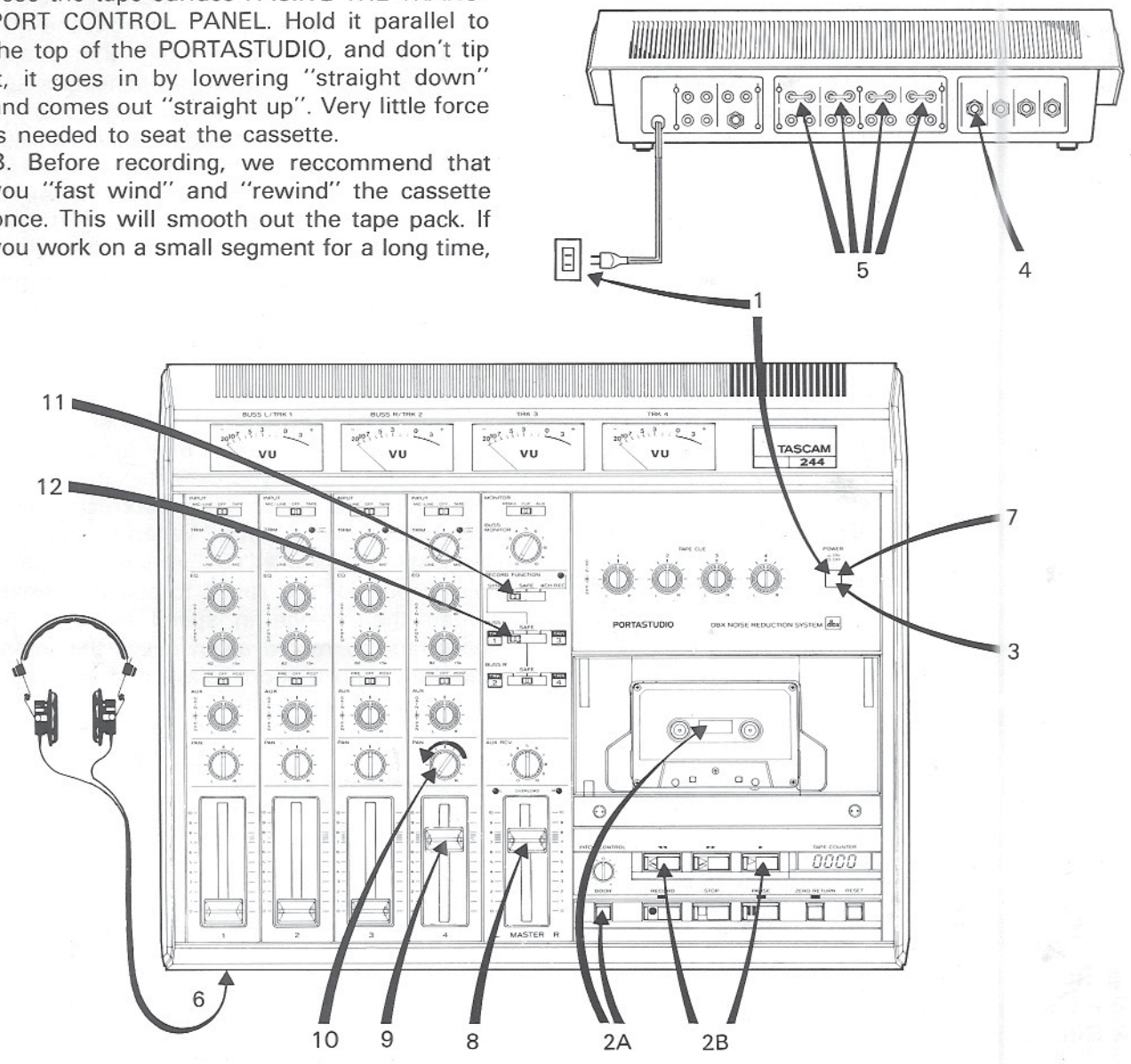
A CAUTION: DBX II calibration will only be accurate at the base speed of 3-3/4 IPS. Recording at one speed and playing back at another, either higher or lower, may cause dbx decoding errors to have an effect on the dynamics of the signal. Since changing the speed of the tape will also alter the tonality and the pitch, the use of this speed shift circuitry will be an artistic judgement that we must leave to you. The published specifications for the unit may not be achievable if this circuit is used to extremes.

This large foldout is intended to provide the necessary instructions for operating the PORTASTUDIO in as brief and compact a form as possible. If you understand what you are being asked to do, all well and good; but if you need more explanation to make an instruction clearer, each line has a page reference at the end of the instruction. .A more complete discussion of the feature will be found on that page.

Recording the First Track

If you are ready to record, we assume that you have the neccessary supplies and equipment.

1. Connect AC power, and turn the PORTASTUDIO "ON". (pg 53, Specs)
2. Load a cassette.
   A. Press DOOR to open compartment. The cassette is loaded with the windows that expose the tape surface FACING THE TRANSPORT CONTROL PANEL. Hold it parallel to the top of the PORTASTUDIO, and don't tip it, it goes in by lowering "straight down" and comes out "straight up". Very little force is needed to seat the cassette.
   B. Before recording, we reccommend that you "fast wind" and "rewind" the cassette once. This will smooth out the tape pack. If you work on a small segment for a long time, say, ten passes on a 2 minute section, do a complete "fast wind" before continuing and you will keep the "pack" smooth and avoid tape troubles. ("Cassette Tapes," pg 15).
3. After winding the cassette, turn off the power while making connections to the unit. We'll tell you when to turn back on.
4. Connect a source of signal. We show Input 4 here. ("Connector," pg 18)
5. Make sure that the ACCESS-SEND-RCV jumpers are in place. Modules won't work without jumpers. ("ACCESS-SEND-RCV," pg 5)
6. Connect stereo headphones only. Mono headphones will damage the unit. Don't put on the headphones until all adjustments are complete. We'll remind you in this list. ("Headphones," pg 44)

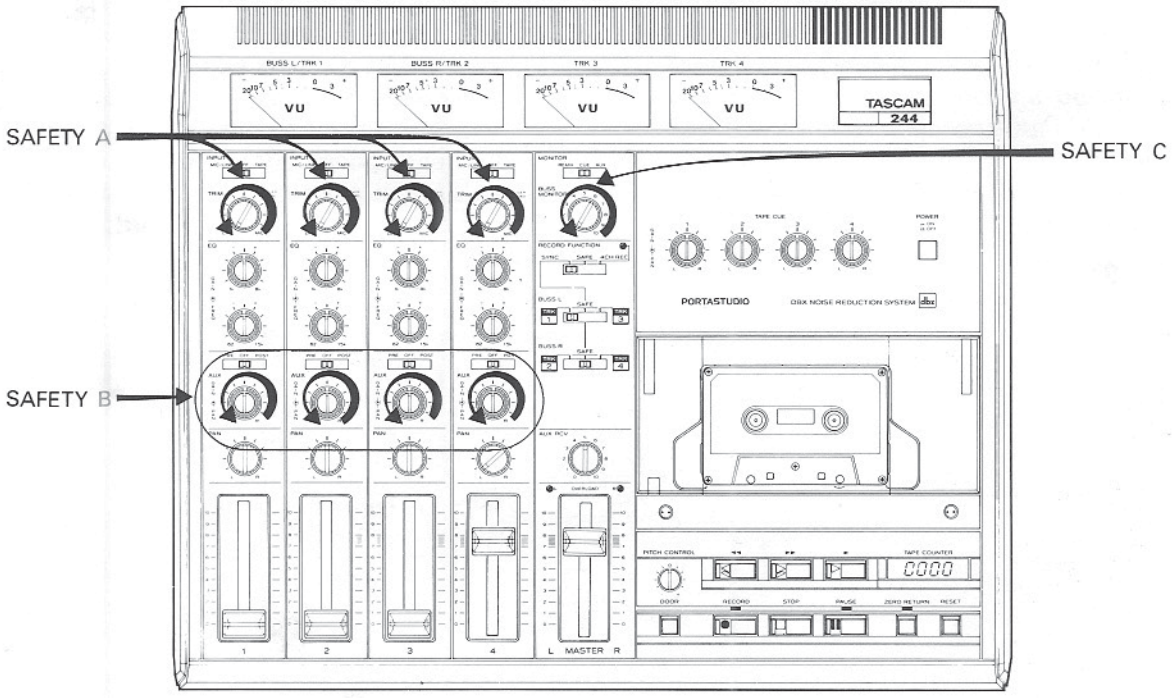


The following three instructions are a safety precaution, and are not strictly necessary. We won't give them numbers, but you'll see the reason for caution if we explain.

SAFETY A: Rotate all 4 module TRIM fully leftwards. This insures that no signal will enter the Module until you are ready to adjust level.

SAFETY B: Set all AUX PRE-OFF-POST switches to OFF, and rotate all AUX-GAINs fully leftwards (off). Even though we are not going to use this section on this first track, if it is working and overloading it may "leak" distorted signal into an active line on the PORTASTUDIO and cause troubles.

SAFETY C: Rotate the BUSS MONITOR control fully leftwards. This action keeps all signal out of the headphone circuit until you are ready to adjust the listening level. Since you have no idea how "loud" your signal is, it is a good idea to start your adjustments from a dead safe "full off" postion, but in later work you may leave a convenient "working level" set on these controls. "zero start" is not mandatory.



7. With all connections made, turn on power.
8. Set L+R BUSS MASTER slightly below the "marks". (pg 27, Parts List).
9. Set #4 Input Fader slightly below marks (pg 28, Proper level setting, pg 22 Parts list.)
10. Set Input PAN fully to the left. (pg 27, Parts List.)
11. Set RECORD FUNCTION top switch to SYNC. DON'T USE 4CH UNTIL YOU READ PG 35 (WARNINGS) (pg 31, Parts List)
12. Set RECORD FUNCTION L BUSS TRACK Select switch to TRACK 1. (pg 32, Parts List). This will light the lamp in Meter 1 and cause the RECORD LED to blink.
13. Set the #4 INPUT switch to MIC/LINE.
14. Set RECORD FUNCTION BUSS R Track Select switch to SAFE. (pg 32, Parts List).
15. While looking at Meter 1, rotate TRIM #4 until a suitable reading is seen, not heard. To separate learning from playing, use a radio as a source of signal, or record a friend. (pg 18, Parts List).
16. Set MONITOR Select switch to REMIX. Now, put on headphones. No sound will be heard until you perform the next setting. (pg 44, "Headphone Amp").

17. Rotate the BUSS MONITOR control (clockwise) to set a listening level. If the signal does not appear in the left earpiece, reverse the headset. The signal will then "track" the action of the Input PAN. After you get the headphones on correctly, switch the MONITOR select switch to CUE (center setting) and you will have a center feed mono signal to work with. For more on this, see pg 31 (24, Parts List) and pg 42 ("Fancy Cues," Overdub Section).
18. Adjust EQ if desired. (pg 23, Parts List).

The Mixer, the Monitor and the Record Function sections have now been completely adjusted, but a little "housekeeping" remains. You must check and set some of the Transport controls before recording.

19. Enable the Zero Return (depress to "set") (pg 34. Parts List).
20. Set a zero on the counter by pressing the RESET button. (pg. 34, Parts List).
21. Check PITCH CONTROL, set for basic 3-3/4 IPS operation at center (detent position). This control is active in both Play and Record. (pg 35, Parts List).

FINALLY

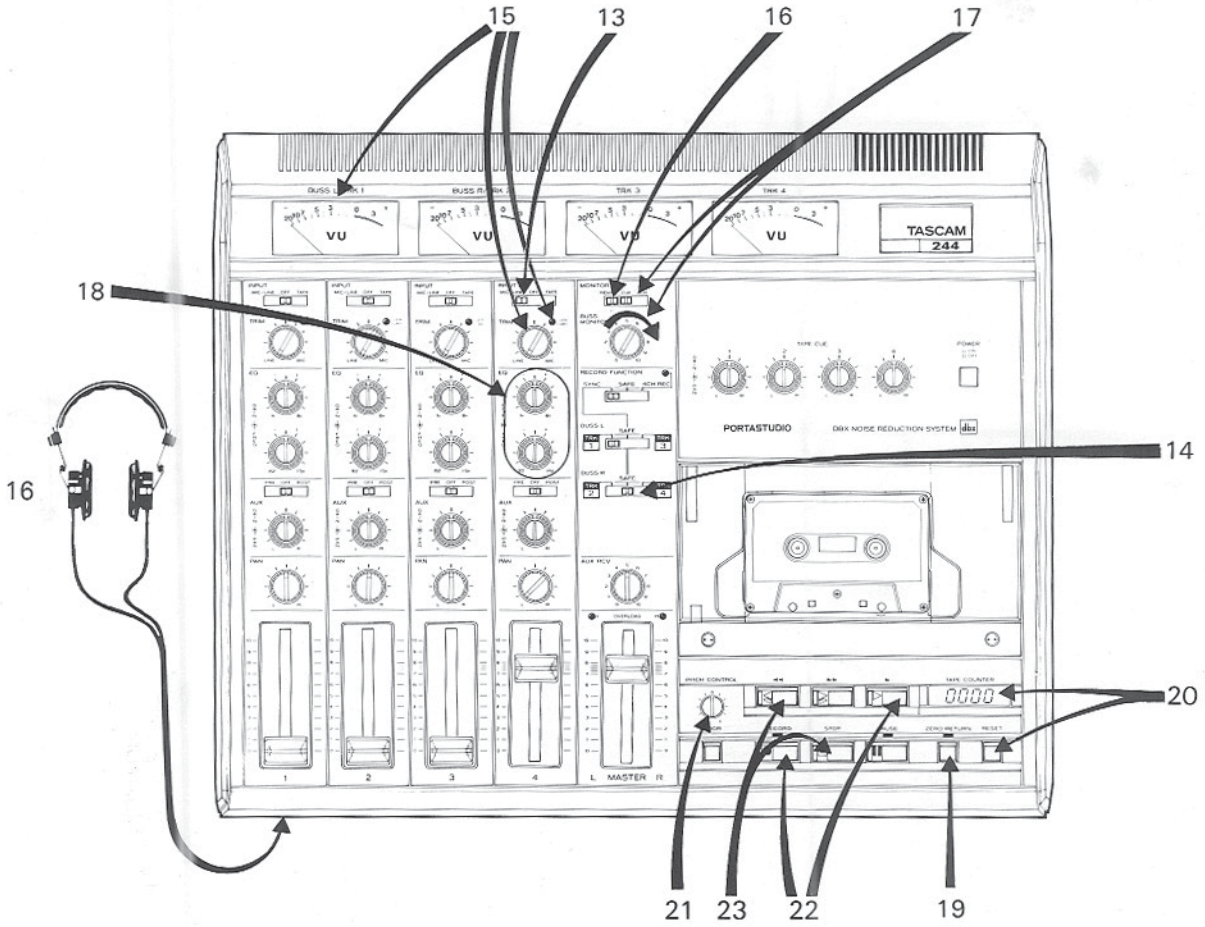
22. Depress RECORD and ▶ (Play) buttons together to start recording. RECORD LED will change from "blinking" to "steadily on" if the record circuits have engaged.

NOTE: If you plan on working alone, the TASCAM RC-30P PUNCH IN/OUT REMOTE PEDAL is available as an accessory.

Connect the RC-30P to the rear of the PORTASTUDIO. Now, even with both of your hands occupied, PUNCH IN/OUT can still be performed by using the remote pedal. While in sync reproduce, pressing the pedal with your foot initiates punch-in of the channels for which RECORD FUNCTION has been selected. Punch-out is done by simply pressing the pedal again.

23. After recording is complete, press STOP, then ◀◀ (Rewind). Since we have enabled the Zero Return feature, the tape will stop automatically at counter 0000.

Turn the page, and we'll review First Playback.



FIRST PLAYBACK

To play back the First Track safely, you must reset several switches.

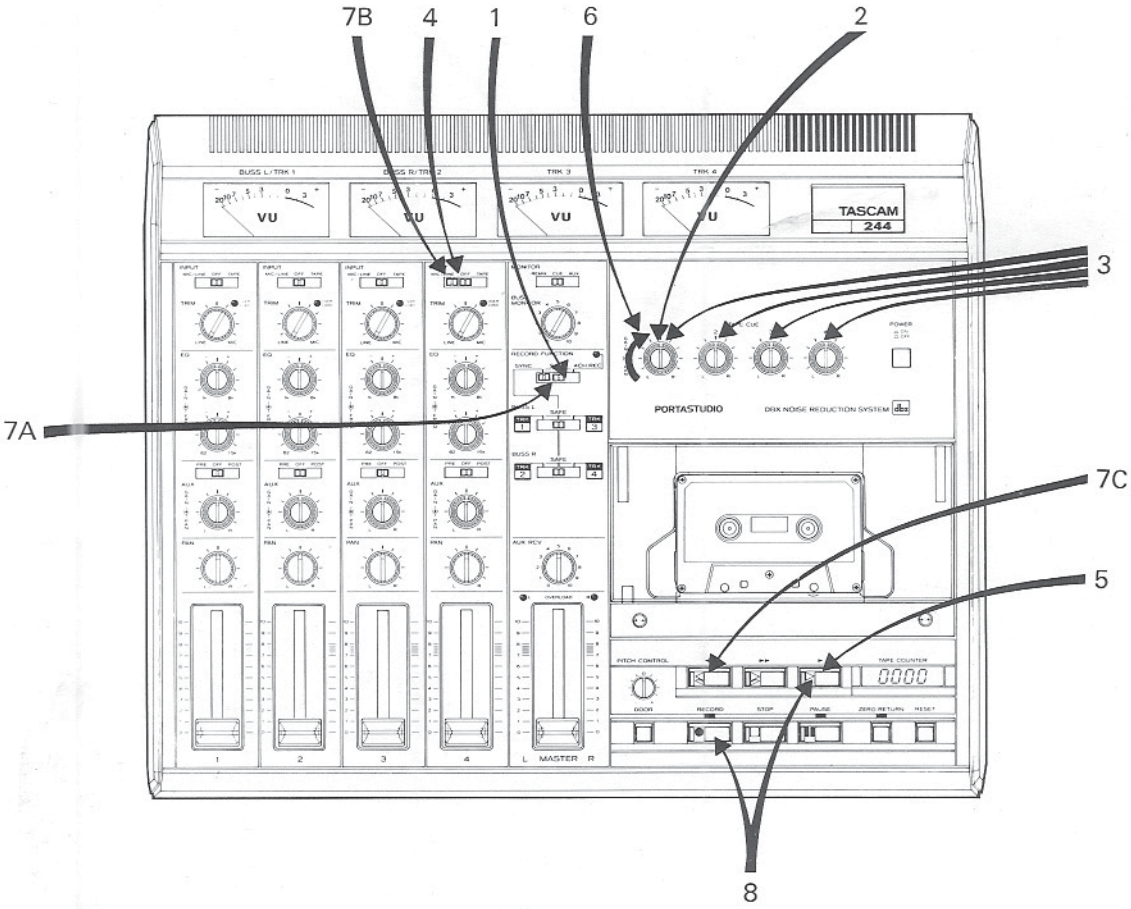
1. Set RECORD FUNCTION top switch to SAFE (Center postion). This will turn off the REC LED in the Transport Control section, showing that you are now "safe". Note that when you do this, ALL 4 METER LIGHTS WILL TURN ON. This is normal, and now indication on Meter 1 will be tape playback level, not mixer output level. You will be able to "see" what you have recorded.
2. Center the TRACK 1 TAPE CUE-PAN (lower section of the dual concentric control, leftmost control in the group above the transport).
3. Rotate all 4 TAPE CUE-GAIN controls fully leftwards (OFF).
4. Switch the #4 Input Module signal select switch to OFF. This will make sure that you don't listen to the mic signal as well as the tape playback. Using this switch as a "mute" allows you to leave all the module controls set for a quick return to recording if you don't like the first "take".
5. PLAY THE TAPE (Press ▶ (Play)).
6. Rotate the #1 TAPE CUE-GAIN control (upper section) to the right until you have a good listening level. Your first track should be playing.
7. If the "take" is not satisfactory, and you want to do it over, only these 3 controls need to be reset.

   A. RE-SET the RECORD FUNCTION top select switch from SAFE to SYNC. (Move it to the left).

   B. RE-SET the #4 Input Module signal select switch to MIC/LINE. Since all the Input Module controls were left set to record, you can make whatever adjustment the first playback seems to suggest is needed. Don't worry about the TAPE CUE controls. They are automatically silenced in the monitor when you reenter RECORD.

   C. When you are ready to re-record, the auto rewind feature will return the tape to the "top". Now----.
8. PRESS RECORD AND ▶ (Play) together and try again.

   When you are satisfied with your first track, we can go on to the next section and "set up" the PORTASTUDIO for the first Overdub.

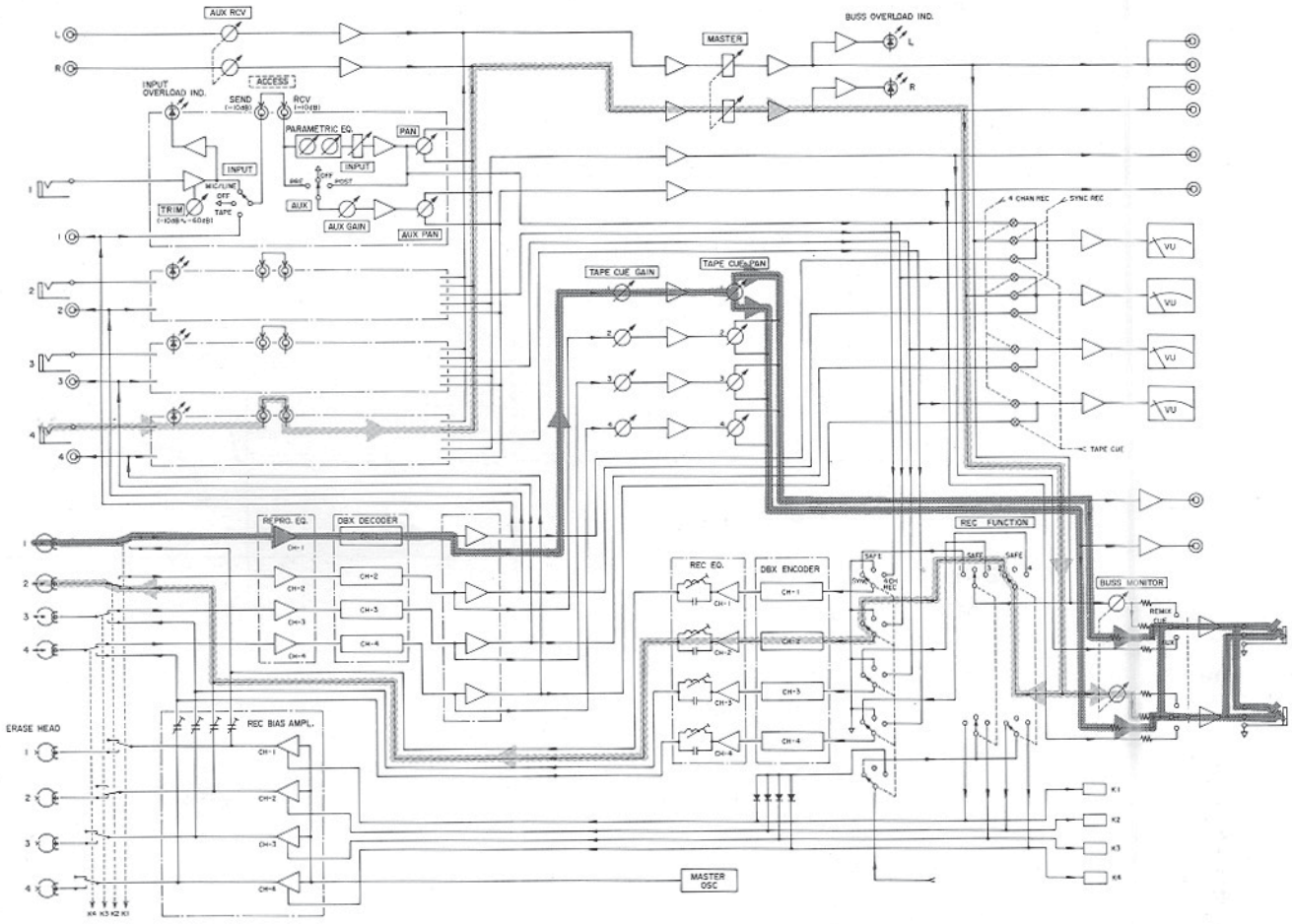


The PORTASTUDIO uses a single tape head for both Record and Playback. You cannot get "out of synchronization" when you record your second part. The Record Track switching takes care of this problem automatically when you select "new" tracks to record on. The only overdub requirement is that you must use the CUE setting on the MONITOR signal select switch. It is the only setting that can mix the Tape Cue signals with the L + R Buss signals. This combination of six mono signal sections will allow you hear everything when you need to balance "old signals" that are on the tape with "new signals" that are on the L + R Buss from the mixer. The 4 Tape Cue sections are always (and only) setting levels for tape tracks that have already been successfully recorded, and never anything else. Is there any other way to get this "two way street" of old and new signal in the headphones? Yes, but before we review any complex methods for constructing a more elaborate cue system, lets make our first attempts with the simplest control system and leave the Fancy Tricks for later. This is the simplest cue system. You do two things:

A. Pick a new track to record on.
B. Balance the headphone signal so you can hear.

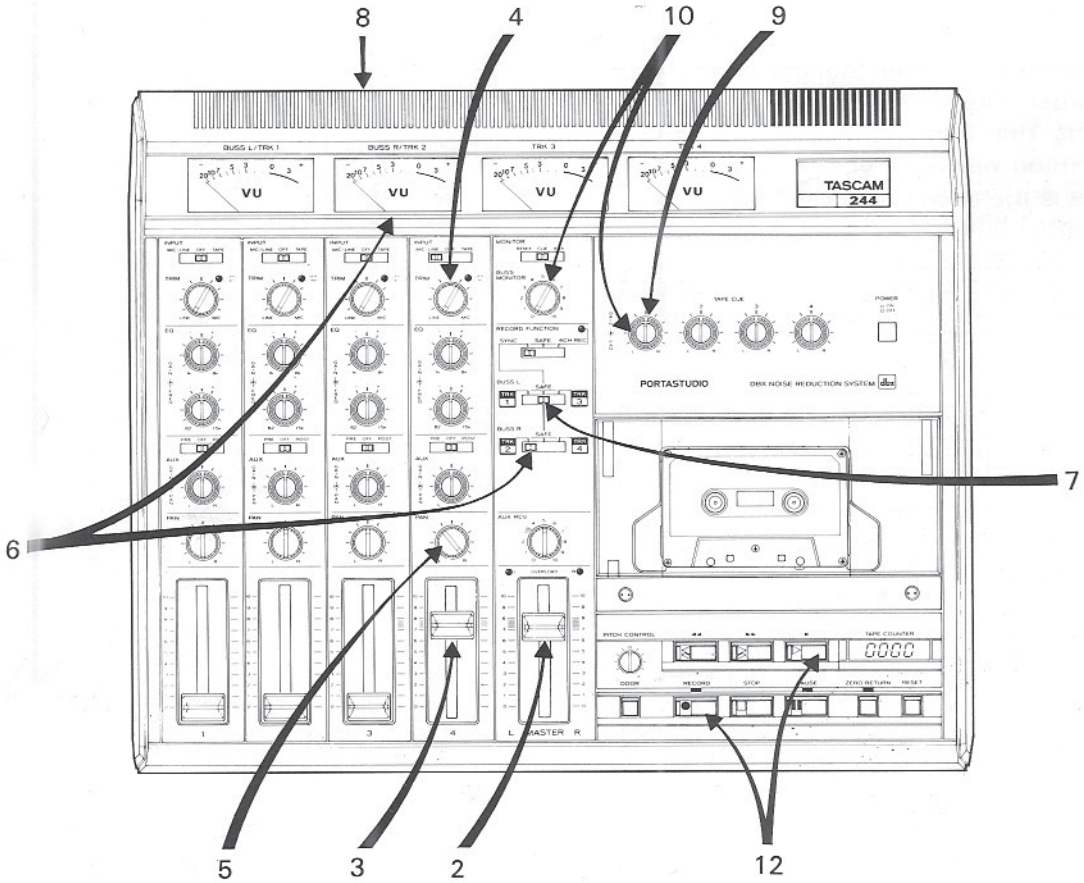
Control Sequence

1. You may continue to use Input Module 4. Since any input module can be assigned to any track with the PAN Pots and the switches in the Function Select Group, there is no need to move the source of signal to another module just to reach another track. The only Input Module control changes you need to make will be the ones needed to suit the artistic requirements of the new part. A new part may require a different EQ, a new instrument may require some experimenting with the TRIM and Fader settings to get an acceptable meter reading.



2. Set the L+R MASTER.
3. Input Fader, and then—
4. Final adjustment of the new signal with the Input TRIM.
5. To get signal to Track 2, first you rotate the #4 Input PAN control fully right. For "one track at a time" overdubs you may leave the Input PAN centered, so it feeds both the L and R Busses equally. This action is not usable when two tracks are recorded, either as "single" or as "true stereo". When two tracks are being recorded you must use the PAN for signal assignment or you will have no isolation and thus, no separate control of each track in mixdown.
6. Set the BUSS R RECORD function select switch (that's the lowest one in the group) to the left, selecting TRACK 2 as the record-ready track. This action lights Meter #2. It is indicating Buss R Output.
7. Set the BUSS L RECORD FUNCTION select switch to SAFE (Center) if you don't have it there already. This action turns off the ready light in Meter #1 and PREVENTS ERASURE OF TRACK 1. Make sure, and avoid disapointment.
8. We assume that the various Cautions about power, ACCESS-SEND-RCV Jacks, and what lights mean danger need not be repeated.
9. We assume that you have an acceptable setting of the TAPE CUE-GAIN 1 and PAN controls, they were just used to check the Track 1 performance.
10. We also assume that you have an acceptable working level set on the BUSS MONITOR Master. This controls only the headphone level of the "new part". The TAPE CUE-GAIN controls the levels of the "old parts".
11. But—before recording, we suggest that you check out the relative levels of the two signals with one (at least) rehearsal. Do you hear both parts clearly? Perhaps a little louder? When the headphone mix is "tuned up", then record.
12. Press RECORD and ▶ (Play) and try the Overdub.

This is where you will spend the long hours with the PORTASTUDIO; in the Overdub modes. The monitor system will be your working tool. This mode of operation will become more familiar to you as you acquire more experience with the unit.



FANCY CUEING

It is not uncommon on a 4 track recording to discover that you have rewound and played the tape 50 or more times before you are ready to "mix". Throughout this process the headphone monitor must provide cues, and a guide to what is happening to the piece artistically as well. Now we must make a "bow" to the art. There are some times when a simple monitor system will not be enough to insure success. You will need help to "squeeze" that last drop of value out of what you hear in order to perform better. Only the most hardened of recording studio professionals can play well when the monitor signal is not ready to "mix" yet. All the corrective measures that are provided have not been used yet, and the signal that you listen to may change dramatically as art when you add more parts to your piece. When our simple cue system cannot solve your problems, we offer these suggestions. They have limits, require sacrifices in mixer function and flexibility, and in some examples, require the use of "outboard" connections to other equipment. Read carefuly.

Tape Cue to Mixer

The Tape Cue section over the transport section has not been reviewed in the Parts List section earlier, because it is better thought of as a part of the recorder. The whole system can only be connected to the Tape Playback amps, and if there is nothing recorded, you have nothing to "mix". Here is the circuit as shown on the block diagram.

When you have something to mix, you can use the stereo L and R TAPE CUE Outputs to route signal back to the mixer externally. First, the final "parts numbers"

29. TAPE CUE-PAN 1-4

These lower sections are pan controls. their action is similar to the Aux pans and the Input pans. Left rotation favors the L TAPE CUE Line, Right rotation sends more signal to the R TAPE CUE Line.

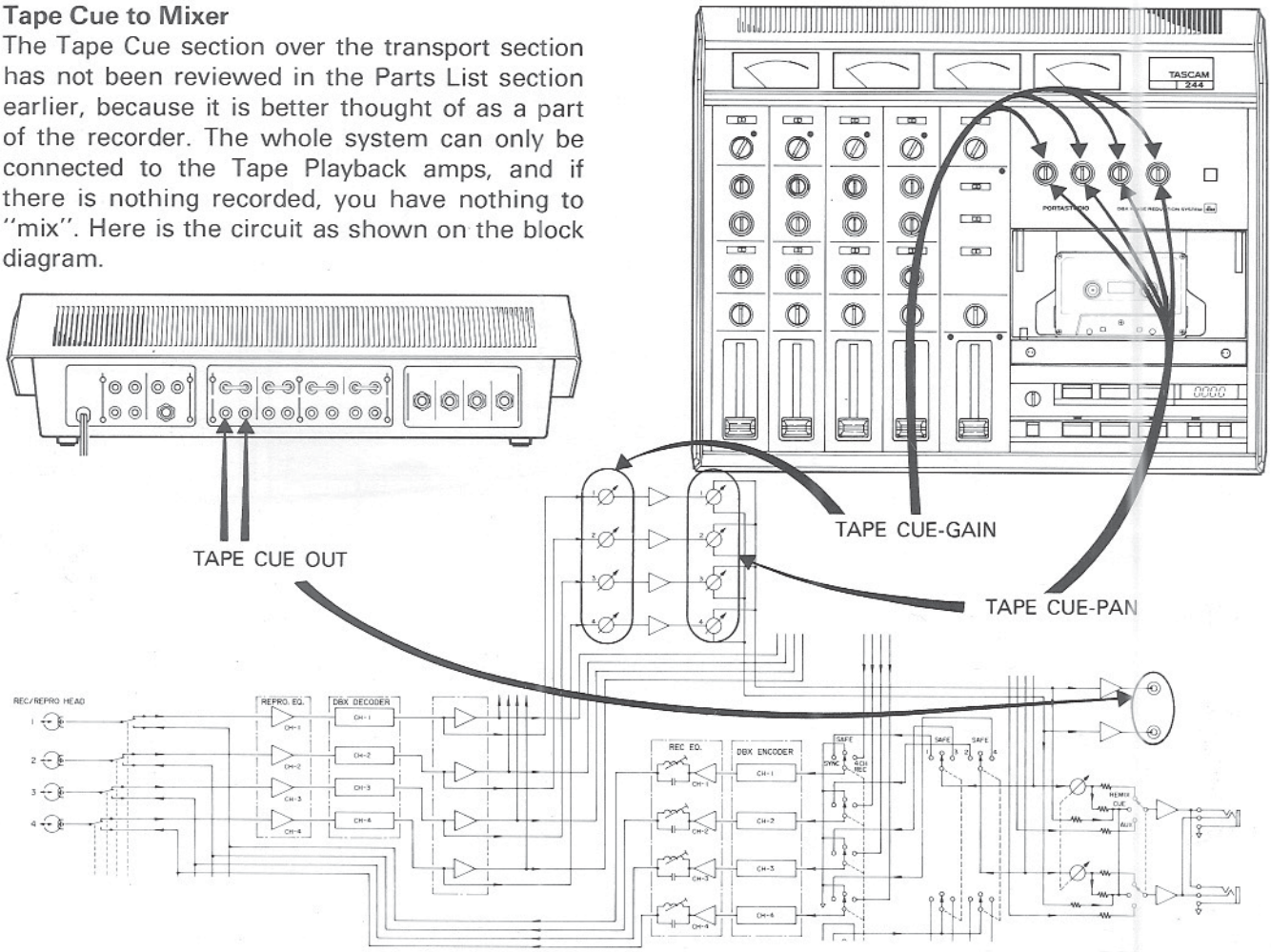
30. TAPE CUE-GAIN 1-4

Provides level control for each tape track, upper section of the dual concentric control.

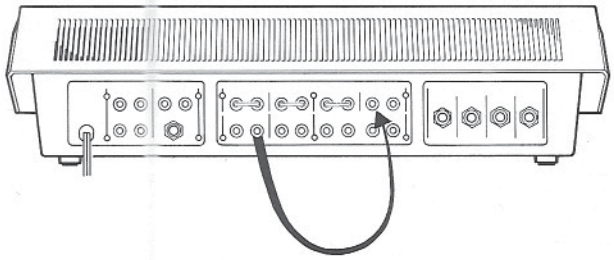
31. L and R TAPE CUE OUT RCA Jacks

The output impedance of these sends is 100 ohms. The nominal level is −10 dB, 0.3 volt. The max. output before clipping is +15 dB, 5.6 volts.

Remember that the internal connection to CUE position of the headphone MONITOR select switch is always "Mono". The TAPE CUE PANs affect only the RCA L + R outputs.



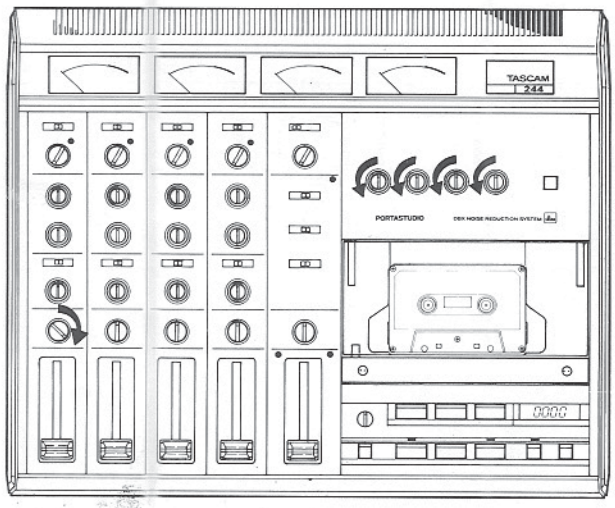
Fancy Cue Number One



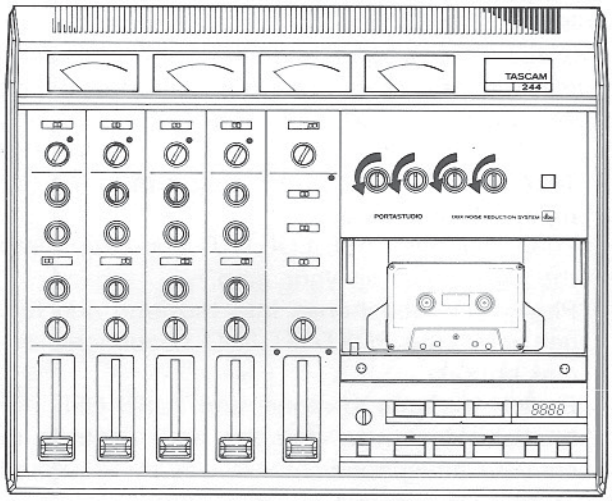
With this connection made, there are assignment possibilities to improve a Cue.

1. Rotate the #1 Input PAN pot to whichever Buss you are not using.
2. The TAPE CUE-PANs should be set full left.
3. Set the MONITOR Signal Select switch to RE-MIX. Since you will be using the Buss to hear the cue, you don't want to hear the Tape Cue Signal in the normal way as well.
4. The remaining 3 input modules can still be used for mics or electric instruments. The price in function of this patch is one input module, and the necessity to remember to reverse the pan on the Cue 1 Input Module every time you change Busses to the recorder. If you forget, you may record the old signal along with the new on the new track. The Aux system is still available for adding effects.

   THE ADVANTAGE: you can now use the EQ controls on the input module to help improve the sound of the Cue signal.



Fancy Cue Number Two



This one uses the same connection in a slightly different way.

1. Input Module 1 is still used as a Cue Master.
2. Leave the TAPE CUE PANs set full left.
3. SHUT THE INPUT FADER ALL THE WAY.
4. Switch the MONITOR signal select switch to AUX. This is the new "sacrifice". You lose the use of the effects system, AUX IS NOW YOUR CUE MIX.
5. Set the #1 INPUT MODULE AUX SWITCH TO "PRE".
6. Set AUX-PAN to "center" the signal in the headphone. This is the benefit: A center mono feed for the headphone circuit that will not need any changing around when you select a different L+R Buss. You lose the EQ, because of the necessary Pre-EQ and Fader Aux routing. There's more to set.
7. On the remaining three input modules, set AUX Signal Select switches to "POST". This will allow you to hear EQ change and fader moves, but—
8. REMEMBER THAT YOU CAN'T HEAR DISTORTION OR "LOW LEVEL" ON THE L+R BUSS! You are listening to AUX, and you will still hear signal even if the L+R MASTER IS SHUT DOWN. The AUX system GAIN and PAN can now be used to set up a "Mix" that can be totally different from the feed to the recorder as regards stereo and gain. AUX-PAN can be set anywhere without causing troubles, and you can make any balance you require out of these 7 signals. To be sure that everything is O.K. on the L+R Buss when you are recording, you can switch the MONITOR signal switch back to REMIX. You will hear the stereo with the L+R BUSS level and Input PAN perspective instead of the AUX-GAIN and AUX-PAN perspective.

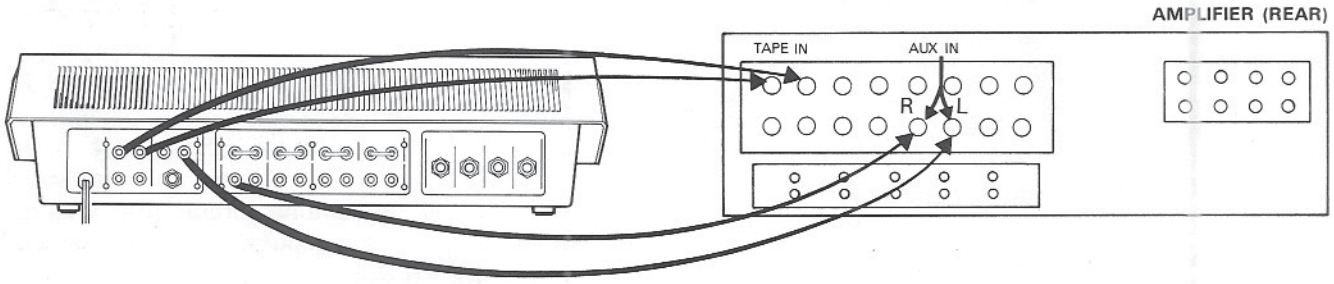
Dirt Cheap Full Mic Cue Without Sacrificing An Input

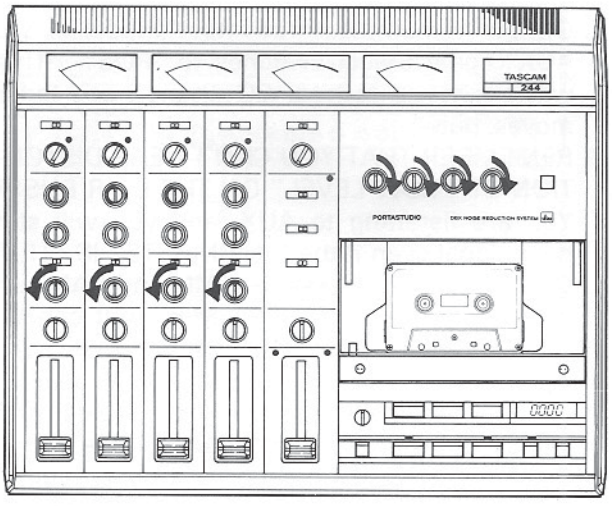
This setup still costs you the normal use of the Aux circuit for effects, and uses your receiver or integrated amp. Do you have an unused line level stereo input on your amp?

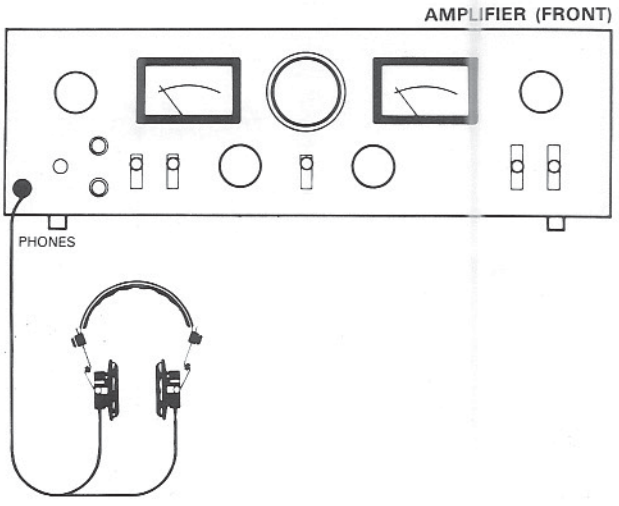
1. Take one AUX SEND and plug it in to the left input on your amp.
2. Then take one TAPE CUE OUT and plug it into the right input on your amp.
3. Plug your headphones into the amp and forget about the PORTASTUDIO headphone circuit entirely.
4. Switch the amp to mono, and TURN OFF The loudspeakers, now use the amp controls to set your headphone volume, and the AUX and TAPE CUE controls to "mix" with.
5. Set all AUX-PANs to the one line that goes to the amp.
6. Set all TAPE CUE-PANs to the one line that goes to the other side of the amp. Now, if you also have the L+R LINE OUTs connected to the amp, the Amp Select switch can be used as a Signal select, and you will have pretty much the same function duplicated on the amp that you started with on the PORTASTUDIO.

32. Dual Tip-Ring-Sleeve HEADPHONE Amp Output

A high power Headphone amp section is provided, and the dual jack allows two people to work at the same time. Max rated power into 8 ohms is 100 milliwatts. Since this is a small power amp, don't attempt to use it to send signal to a low level input without using a resistive pad. However, if you have a set of small high efficiency loudspeakers with an input rating of 8 ohms or better (a larger number for ohms) and a rated efficiency of better than 92 dB/1 meter, 1 watt, they may prove to be usable! Take care, they will not play as loud as a studio monitor.

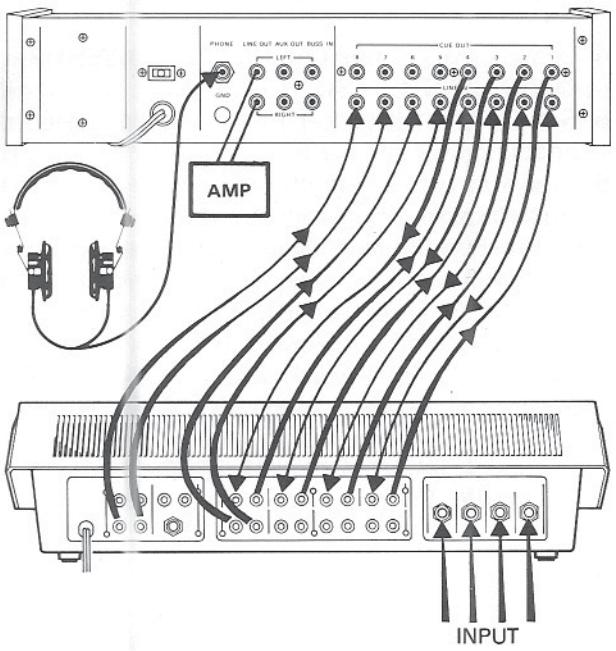






Using a Model 1 To Get a Second Deluxe Cue System

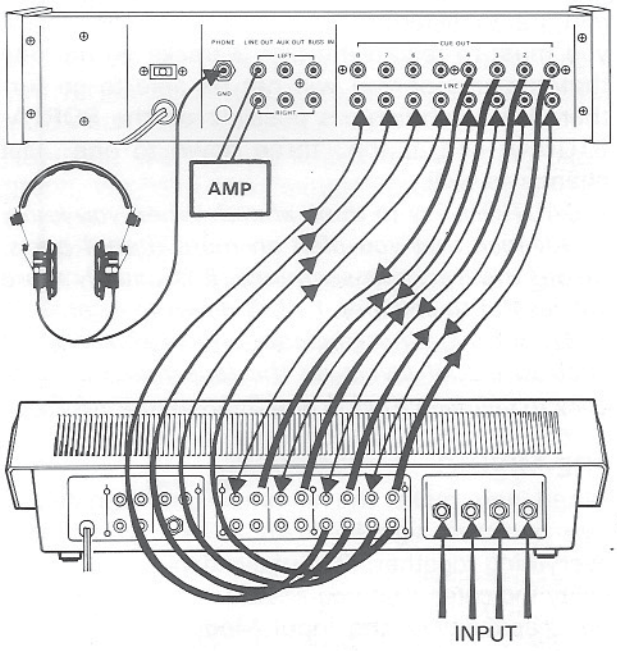
The purchase of our Model 1 Line Level Accessory Mixer will allow you to set up this deluxe Cue. The circuits already provided may be bypassed, or used along with this outboard Model 1.



1. Connect the first 4 lines on the Model 1 to the ACCESS-SEND-RCV Points on the PORTASTUDIO.
   This connection allows for a separate Input Cue mix that can be arranged differently than the mix in the PORTASTUDIO.
2. Connect the TAPE CUE L and R OUTs to Model 1 Line Ins 5 and 6. This mix will be the same as the mix on the PORTASTUDIO Tape Cue system.
3. Connect the L + R Buss Spare (AUX) Line Outs to Model 1 Line Ins 7 and 8. This connection will allow you to hear the same mix as the REMIX setting on the PORTASTUDIO (with possible effects returns as they are being recorded).

Second Patch For Model 1 Addition

If a totally seperate Cue system is desired, rewire this way:



1. Connect the first four model 1 Line Ins (1-4) across the ACCESS-SEND-RCV Jacks on the PORTASTUDIO.
2. Connect the last four Model 1 Line Ins (5-8) to the TAPE OUTs, 1 thru 4 on the PORTASTUDIO.
   This patch gets all 8 signals to the Model 1 as a pre-everything feed from the PORTASTUDIO. Now, any mix made on the Model 1 will be totally independent of all control settings on the PORTASTUDIO except the TRIM on the 4 Input module signals.

HOW TO MAKE A "PING PONG" (COLLAPSE, ALSO CALLED "BOUNCING TRACKS")

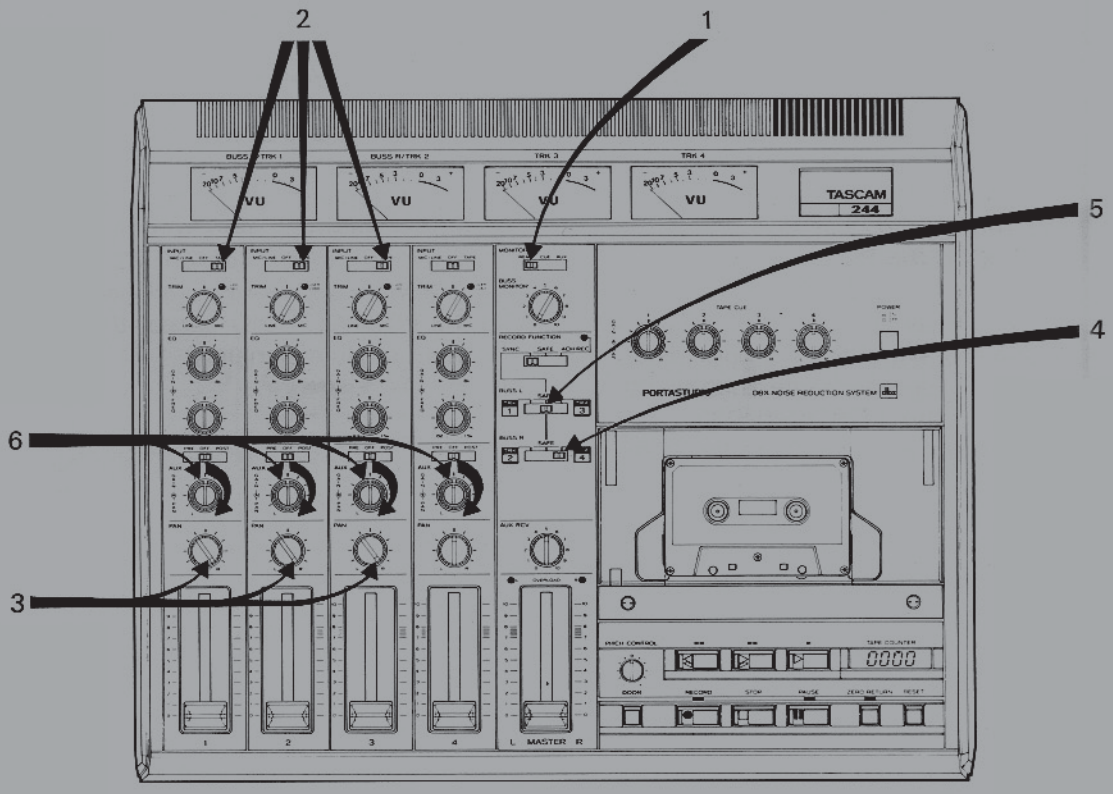
The availability of 4 tape tracks offers a lot, but sooner or later, you will begin a piece that requires "just one more part". It's inevitable, and the "one more" soon expands to strain the technology, and we now think nothing of 24 tracks. How does this affect the PORTASTUDIO? When you are considering the addition of another part, you must re-record the first 3 tracks on the one that remains or you will not be able to go further. This technique is possible on the PORTASTUDIO, and at the "three down to one" last chance as well.

A CAUTION: Try to think ahead. When you know in advance that you plan on more than 4 parts, record the tracks in sequence. It is usually more successful to collapse 1, 2, 3 down to 4, or 4, 3, 2 down to 1. Unpredictable effects that are related to actual levels on the tape make it more risky to collapse track 1 and 3 down to track 2.

PRE-MIXING

When you make the all important transfer of two or three signals down to one, you "lock" everything together. This unavoidable loss of flexibility indicates that you should use the full function capability of the Input Modules. Here's the setup.

1. Set MONITOR Select switch to REMIX.
2. Set Input Modules so all 3 receive Tape signal with INPUT Select (set Right).
   We show 1, 2, 3 set to record on 4.
3. Set all three Input PANs to BUSS R (PAN FULL RIGHT). R Buss is the one that can record Track 4
4. Enable Track 4 record by setting LOWEST RECORD FUNCTION SWITCH to the right.
5. Set "SAFE" (Center) on the middle record enabling switch.
6. Any Aux system effect must return to Buss R Aux Receive or it will not be recorded. Remember, you have only one line, That's mono.
7. *A Final Note: You may add a fourth signal to your "ping-pong" by using Input Module 4. For most first time users the fact that THE INPUT 4 IS NOT THE TAPE TRACK 4 is very hard to remember. It simply "looks wrong". We say, look again, and you will see that it will work. INPUT 4 IS NOT USED OR BLOCKED WHEN YOU RECORD TRACK 4.*



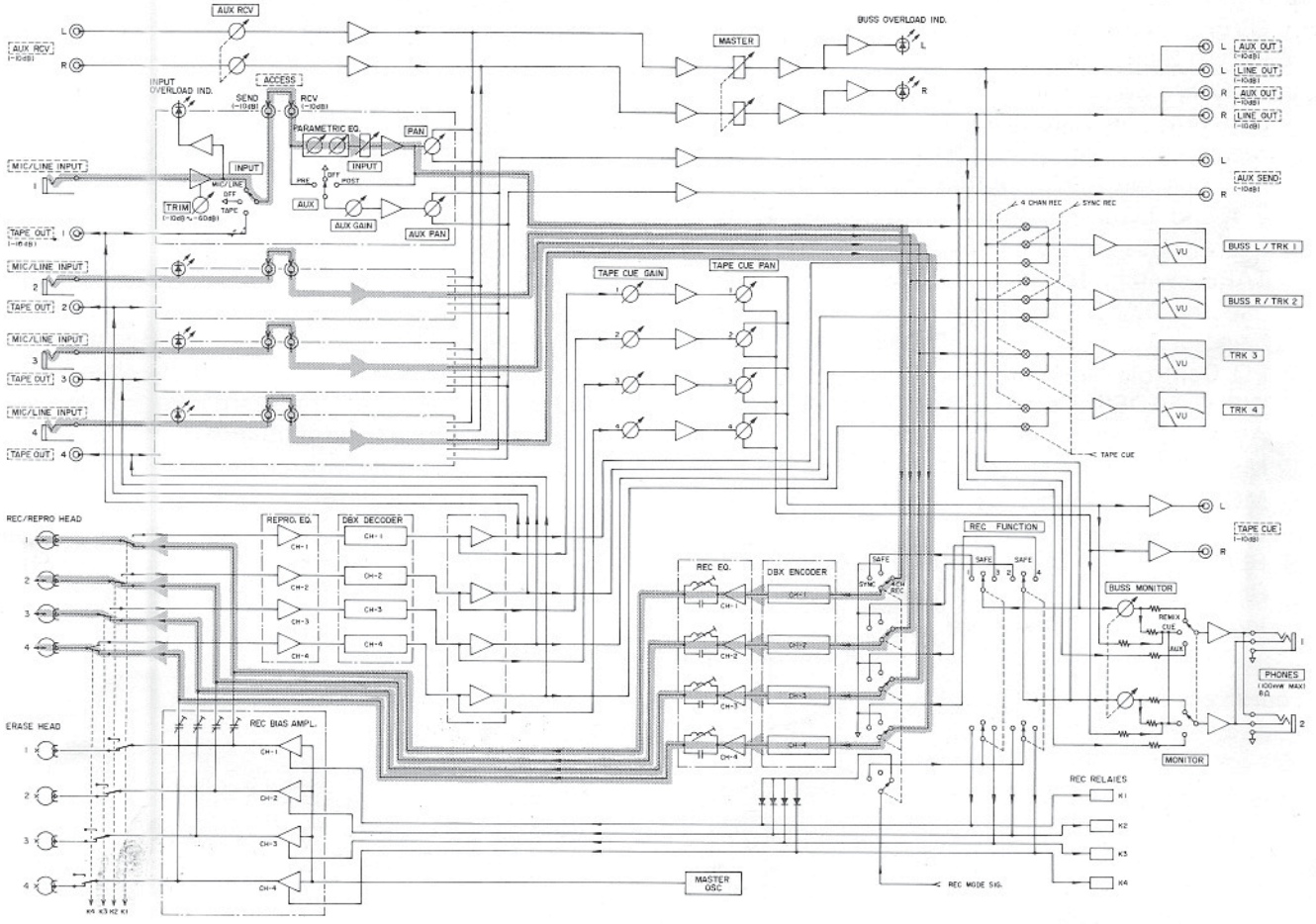
FOUR CHANNEL RECORD SPECIAL FUNCTION AND THE METER 4 CHANNEL RECORD LOGIC

When the Four Channel Record function is selected, each input module is assigned to its corresponding track on the recorder, and the four meters now read the signal derived from this point in the wiring, Module 1 goes to Meter 1, and so forth. Although the PAN Pots still feed the L+R MASTER FADER, it is no longer connected to the recorder. You can still hear "stereo" in the Monitor, and the level will be affected by the position of the L + R MASTER, but to the recorder, no "mixing" is possible. You will now have one module feeding each track, and the recorder will ALWAYS ERASE ALL FOUR TRACKS PRIOR TO RECORDING ALL FOUR TRACKS SIMULTANEOUSLY! YOU CANNOT MAKE AN OVERDUB IN THIS MODE. The Tape Cue circuit is disabled, depriving you of the necessary headphone feed, and anything that is already on the tape will be erased when you start 4 Ch Rec, so take care. This special function is provided to allow copies to be made, and to record all four tracks "from scratch". It is definitely not the "normal" method of use. True, it is possible to begin a tape in this mode, you can record one, two, or three tracks at one "pass" with this logic, switch to SYNC (That is the Overdub mode) and finish the remaining track or tracks but if you forget to switch from 4CH REC to SYNC, YOU WILL ERASE YOUR FIRST "PASS".

Now that we have re-stated the warning paragraph, we have two "setups" that you are likely to need.

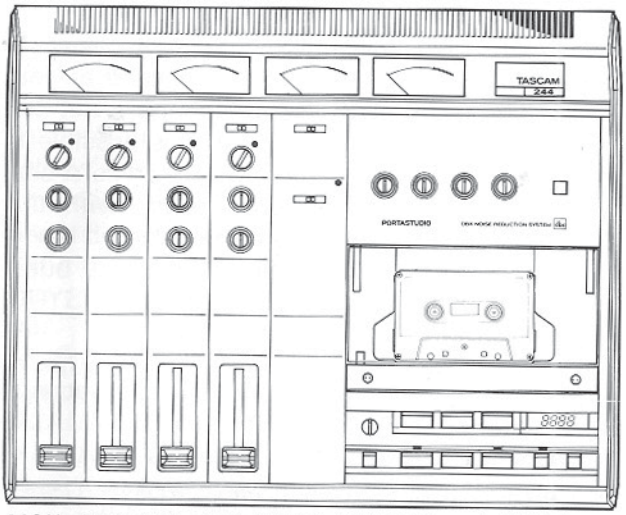
1. COPYING.
2. DISCRETE 4 CHANNEL RECORDING USING MICS.

Let's start by reducing the block diagram to the lines that will actually reach the tracks on the recorder.



As you can see, this is a pretty drastic reduction. Most of the controls that have been omitted will still function, but—they no longer affect the recording. For those of you who are still a little uncomfortable with reading block diagrams, we offer this illustration of the top panel. It shows the active controls for Copy or 4Ch mode on the next page.

Controls that affect 4CH



MON, AUX, CUE, AUX RCV, L+R BUSS work BUT do not affect 4CH record to tape. To check playback, use TAPE CUE.

THESE CONTROLS ARE STILL ACTIVE.

1. 1/4 inch PHONE INPUT Jack. Normal.
2. TRIM. Normal operation.
3. OVERLOAD LED. Normal.
4. INPUT Signal Select Switch. Normal.
5. ACCESS SEND-RCV. Normal. If you are certain that the output of the 4 channel Play recorder or system is −10 dB (0.3 volt), you may use the RCV as a line input when copying and bypass the first gain stage. Doing this bypass will disconnect the MIC/LINE-TAPE switch. To "check" your copy quickly, use the TAPE CUE System when you play back.
6. EQ Controls. Normal.
7. INPUT FADER. CHANGED, NOW OPERATES AS TRACK RECORD MASTER.
8. METERS. CHANGED, NOW READ INPUT MODULES 1-4, USE AS TRACK 1-4 RECORD LEVEL INDICATOR.
9. RECORD FUNCTION SWITCH. SET TO 4CH, REC causes the actions that implement the logic we have listed here.
10. TAPE OUT 1-4. Normal. Use to feed another 4 Track recorder when this PORTASTUDIO is the Play recorder. Since there is no level control on this output, level adjust of this signal will have to be done on the recorder that is recording the copy.

To Monitor a 4Ch Recording.

Use the Input PANs to feed the L+R Buss MASTER Fader. Set the MONITOR Signal Select switch to REMIX. No, the L+R MASTER will not affect the recording, but you may use it to "audition" the 4 signals in order to make "touch-up EQ" and keep track of the sound. The Input Faders 1-4 will control the level seen on the Meters 1-4 and thus, the level recorded on each track.

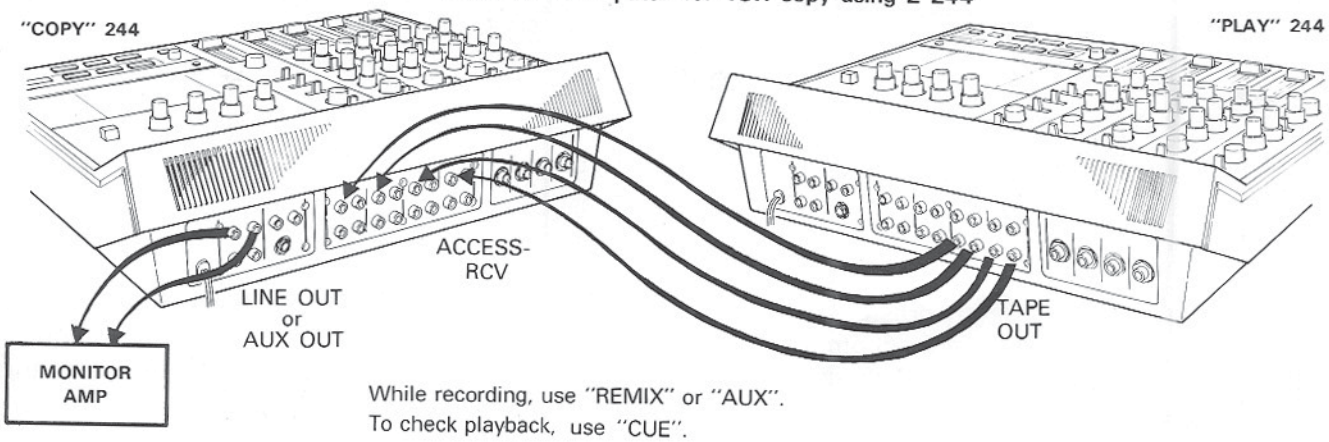
YOU WILL NOT BE ABLE TO MONITOR "OFF THE TAPE" WHILE THE RECORDING IS IN PROGRESS. The PORTASTUDIO only has one head for both the record and play functions, and it is now being used to record.

If the "mix" that appears on the L+R BUSS is not acceptable, you may use the AUX controls on each input module. You can make up a monitor mix that will allow you to hear one part or parts at a more realistic level without compromising the recording quality. Set the MONITOR select switch to AUX.

To Check The Recording.

Rewind, set SAFE, and use the Tape Cue system. Why not use the Mixer Tape Playback function (switch the Inputs to TAPE)? Because you will have to use the Input Faders to adjust levels, and if you have set any corrective EQ you will have to center all the adjustments to avoid EQ-ing "TWICE". This can confuse your judgement of the copy recording if you forget, so use the Tape Cue circuit, and you will be able to hold all your "settings" until you are sure that your copy has been successfully recorded.

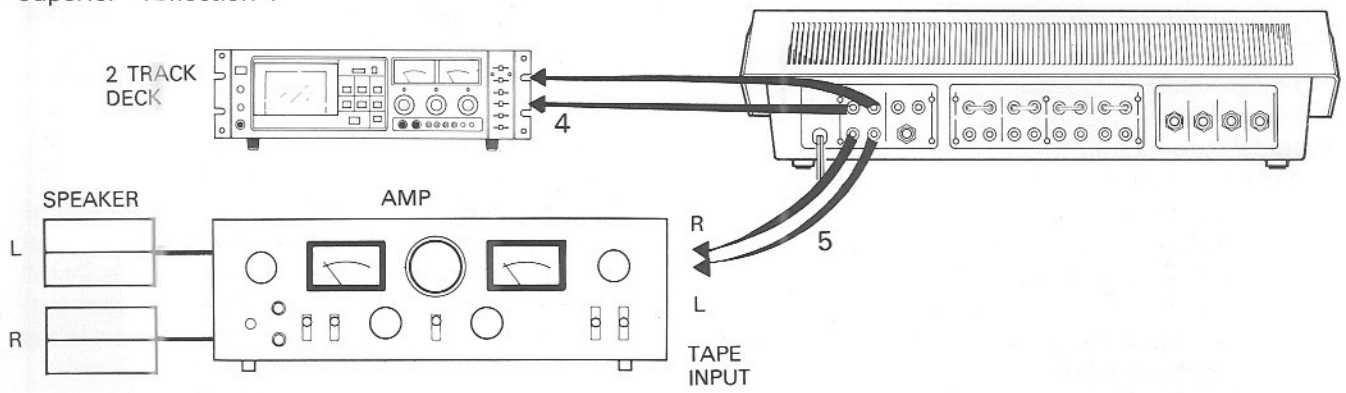
Recommended patch for 4CH copy using 2 244

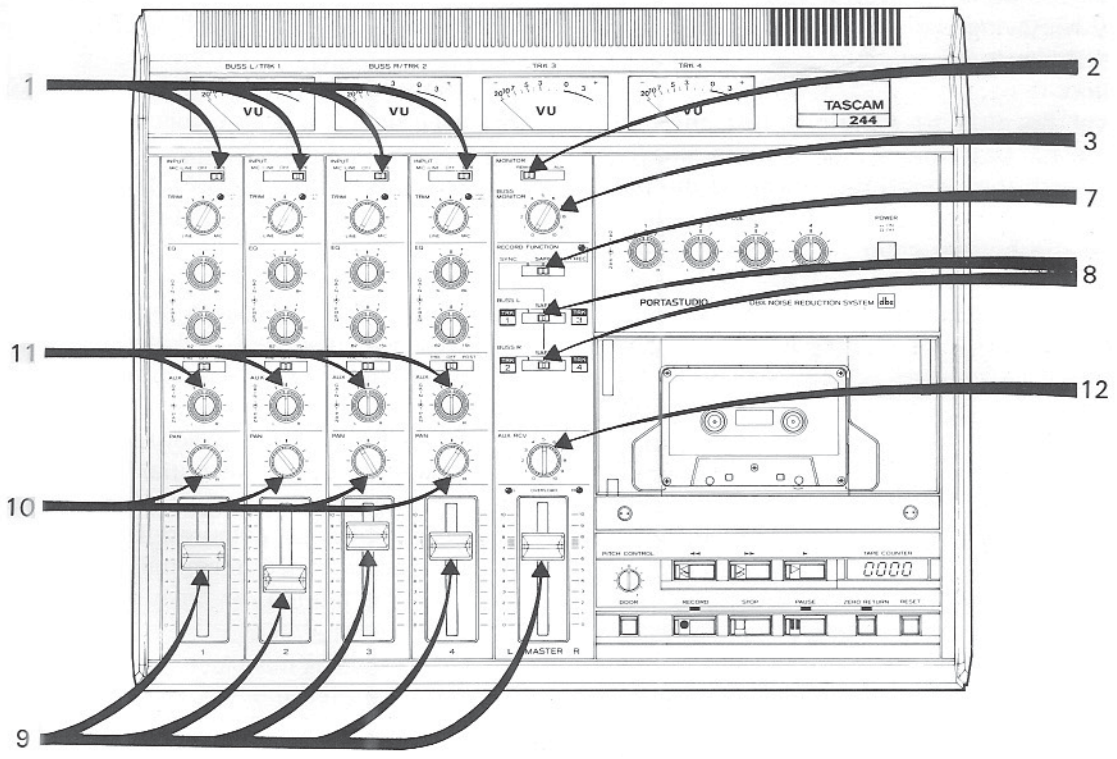


While recording, use "REMIX" or "AUX".
To check playback, use "CUE".

When you have decided that you have done the best you can with your tape, it will be time to mix down to stereo or mono. Since there is only one recorder in the PORTASTUDIO, you must use the LINE OUT to reach another recorder and perhaps a loudspeaker monitor system.
Before the setup, a final word of advice.
Mixdown is the cutting edge of the Multichannel art. This final step in the multitrack process is where you must "face the music" in both the artistic and the technical areas. It is not always pleasant to finally find out that a hard to perform part is just not going to get any better. More work as an artist must come first, and the PORTASTUDIO cannot record what you didn't play. The recorder is only a mirror of reality. Hold up a better "picture" and you will see a superior "reflection".

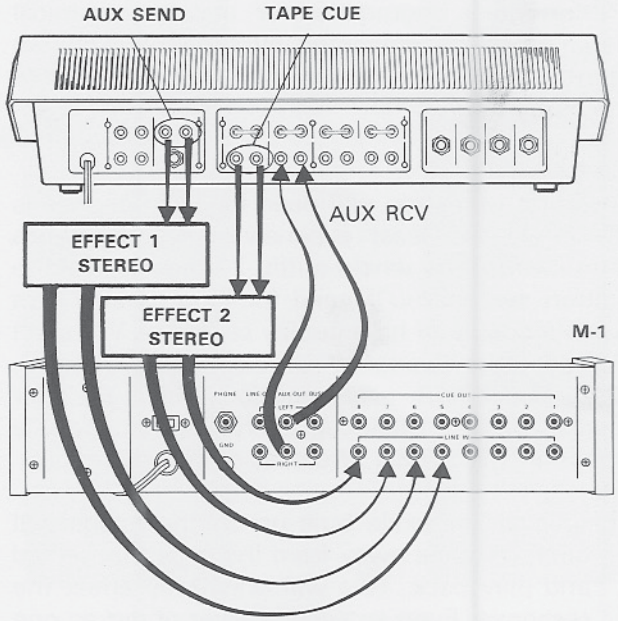
The remaining "tricks" and our previous "fancy setups" will not be capable of improving your performance except in subtle ways. Processing devices such as Echo, Flangers, Delays and the like are useful tools, but no processor has ever been known to move a note to a more timely or tasteful location. Sorry, but that's the truth.
However — Practice and experience will help with the PORTASTUDIO as well as with your art. Give yourself time to understand and become familiar with the technology. Your first tapes will be learning experiences as well as art and you will get better as an Artist/Engineer if you keep at it. Hang in there and it will improve. Have the courage to start over again on your first piece and see if it doesn't "mix" better the second time around. We're sure that it will.





Remix Setup

1. Set all Mic/Line switches on the Input modules to TAPE.
2. If you have no loudspeakers, set the headphone circuit to REMIX.
3. BUSS MONITOR is now your listening level master control, separate from the "send" the 2 track recorder.
4. Connect your 2 track Inputs to the LINE OUT L + R Jacks on the PORTASTUDIO.
5. If a receiver is available, connect the AUX L+R OUTs to an available Line In.
6. If a receiver is available, use the volume control there to adjust loudspeaker level separately from the record level sent to the 2 track.
7. Set 4CH RECORD FUNCTION Select switch to SAFE. How much time have you invested in this tape up to now?
8. Now, the two lower Function Select switches may be safely disregarded. Since the top switch forces "all safe" you will not run any risk of accidentally erasing your master.
9. Level setting for mixdown should start with the L + R Buss MASTER as before, but only the INPUT FADER IS ACTIVE thereafter. Since the TRIM is bypassed in this mode, it is not used and the overload indicator won't work.
10. Use the Input PANs to control the stereo perspective.
11. The AUX System GAIN Controls will allow you to feed a signal out ot a processing device either with or without EQ and a different stereo perspective to the processor is possible by setting the AUX-PAN to a setting that does not match the main Input PAN.
12. The AUX RCV is used to return the signal to the mix after it has been processed.
13. The Tape Cue circuit can be used to feed another device, but no input to the mix is left. Here is a setup that uses a model 1 to combine two returns so they can be patched to the one available AUX RCV on the L+R Buss.



14. Remember that an accessory device may be applied to an individual input by using the ACCESS-SEND-RCV patch point. If you need a processor on only one signal, you don't need another complete mixing system.
15. In addition to the systems described, there still is one possible source for yet another "mix" but it has no controls at all and you sill have to provide everything. the TAPE OUTs are active, but "SEND", "PAN", "GAIN" and "RCV" must all be provided with outboard gear such as our Model 1.

A LAST COMMENT

We can get you set up to mix but here we must wish you luck and leave you to cope by yourself. We don't know what is on your master, or what you intended when you began. Our last piece of advice is: judge your final EQ settings on an individual signal when it is "in the mix". The other sounds will effect the tone you have set in "sold" so be prepared for some corrections. Don't mix on equipment that is grossly different from the tape. What does that mean? consider an audio visual "show". If you prepare on one set of loudspeakers and present on something totally different, you may get a rude surprise. Take care, and we wish you—

GOOD LUCK WITH YOUR SOUND!

Performance degradation or electromechanical failure can be prevented by scheduled checks and maintenance. Periodically follow the check items below:

CLEANING

The first thing you will need for maintenance is definitely the least expensive—cleaning fluids and swabs. The whole outfit, 2 fluids and all the cotton swabs you'll need for months cost less than a couple of high quality cassettes. We can't stress the importance of cleaning too much. Clean up before every session. Clean up after every session. Clean up every time you take a break in the middle of a session (we're serious). Here's why:

1. Any dirt or oxide built-up on the heads will force the tape away from the gaps that record and play back. This will drastically affect the response. Even so small a layer of dirt as one thousandth of an inch will result in a degraded performance. All the money you have paid for high performance will be wiped out by a bit of oxide. Wipe it off with head cleaner and get back to normal.
2. Tape and tape oxide act very much the same as fine sandpaper. The combination will grind down the tape path in time. If you don't clean off this abrasive material on a regular basis, the wear will become irregular. Even wear on heads can be compensated for by electronic adjustments for a time, but uneven wear can produce notches on heads and guides that will cause the tape to "skew" and skip around from one path to another, making adjustment impossible. This ragged pathway chews up the tape, producing more abrasive material, thus causing more uneven wear and so begins a vicious circle that can't be stopped once it gets a good start. The only solution will then be to replace not only the heads, but all the tape guides as well. Being conscientious about cleaning the tape path on the PORTASTUDIO will more than double the service life of the head assembly.

Cleaning the heads and tape guides

All heads and metal parts in the tape path must be cleaned after 6 hours of operation or before starting or after ending a recording session. The TEAC HC-1 head cleaner is recommended.

Cleaning the pinch roller

Clean this at least once each day the deck is used. The TEAC RC-1 rubber cleaner is recommended.

1. Open the cassette door cover.
2. Activate the Play mode.
3. Apply a small amount of RC-1 to the end of a cotton swab.
4. Lightly press the cotton swab onto the pinch roller to the right side of the capstan shaft. This will prevent the swab from becoming entangled. Clean until there is no more visible residue on the pinch roller.
5. Using a clean cotton swab, wipe off all excess RC-1 from the pinch roller. Make certain that there is no foreign matter remaining on either the pinch roller or capstan shaft.

Cleaning the capstan shaft

The HC-1 head cleaner is recommended. After cleaning the pinch roller, it is necessary to clean the capstan shaft.

To do so:

1. Activate the PAUSE mode.
2. Apply HC-1 to the end of a cotton swab.
3. Lightly press the cotton swab onto the rotating capstan shaft.

DEGAUSSING (DEMAGNETIZING)

A little stray magnetism goes a long way. A long way towards making trouble for your tapes that is. It only takes a small amount (1.2 gauss) to cause trouble on the record head and playing 10 cassettes will put about that much charge on the heads and other ferrous parts of the tape path. A little more than that (.7 gauss) will start to erase high-frequency signals on previously-recorded tapes. Demagnetize the whole tape path. This is a standard guideline even though it may be a bit hard to keep track of. Fast motion isn't as significant to the heads, so we don't give an hourly reference. It's the record/play time that counts.

Degaussing is always done with the recorder turned off. If you try it with the electronics on, the 60-cycle current pulses produced by the degausser will look just like 60 Hz audio signals to the heads, at about 10,000 G and will seriously damage the electronics and/or the meters.

Turn off the machine, turn on the degausser at least 3 feet away from the recorder.

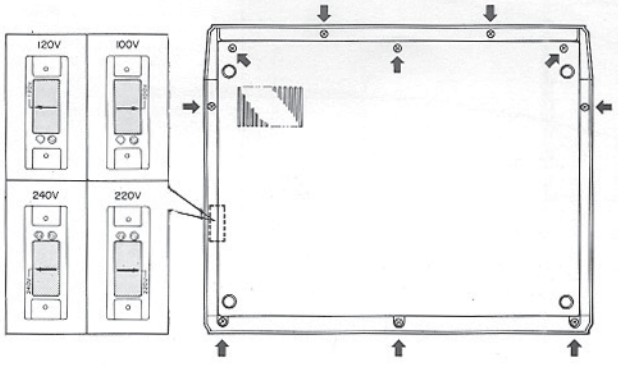
Be certain that your degausser has either a plastic cover or plastic tape covering the tip. Make sure that no metal ever touches the tape heads as it will scar the head and cause it to be unserviceable. Move slowly in to the tape path.

Move the degausser slowly back and forth touching lightly all metal parts in the tape path and slowly move away to at least 3 feet before turning off.
It's a good idea to concentrate when you are degaussing. Don't try to hold a conversation or think of anything else but the job you are doing. If the degausser is turned off or on by accident while it is near the heads, you may put a permanent charge on them that no amount of careful degaussing will remove - head replacement time again, we're sorry to say. Make sure you are wide awake for this procedure.
A clean and properly demagnetized tape recorder will maintain its performance without any other attention for quite some time. Even if it does drift as a recorder, it won't ruin previously-recorded material, and getting it back in good shape will not be too difficult.

Voltage Conversion

General export model only:
If it is necessary to alter the AC power setting to suit the line voltage in your area, use the following procedures:
ALWAYS DISCONNECT THE POWER LINE CORD BEFORE MAKING THESE ADJUSTMENTS!

1. Remove the bottom cover of the PORTASTUDIO by removing the screws.
2. Locate the voltage selector shown in the illustration.
3. Remove the plug by pulling it out, then reinsert it so that the arrow on the plug is parallel with the white line indicating the proper voltage.
4. Replace the bottom cover.



Note for U.K. Customers

U.K. customers only: Due to the variety of plugs being used in the U.K., this unit is sold without an AC plug. Please request your dealer to install the correct plug to match the mains power outlet where your unit will be used as per these instructions.

IMPORTANT

The wires in this mains lead are coloured in accordance with the following code:

BLUE:	**NEUTRAL**
BROWN:	**LIVE**

As the colours of the wires in the mains lead of this apparatus may not correspond with the coloured makings identifying the terminals in your plug, proceed as follows.

The wire which is coloured BLUE must be connected to the terminal which is marked with the letter N or coloured BLACK. The wire which is coloured BROWN must be connected to the terminal which is marked with the letter L or coloured RED.

This product is manufactured to comply with the radio interference of EEC directive "82/499/EEC."

SPECIFICATIONS

MECHANICAL

Tape:	Compact Cassette, C-60 or C-90, Use a gamma-ferric oxide tape that requires high-bias level (chrome position) and 70-microsecond EQ. (TDK-SA, MAXELL-UDXL-II, or equivalent)
Track Format:	4-Track, One Direction (Special Format)
Heads:	4 Channel Erase (Ferrite) 4 Channel Record/Playback (Permalloy)
Tape Speed:	3-3/4 ips. ±1%
Pitch Control:	±15% of normal tape speed
Wow & Flutter:	±0.06% peak (DIN/IEC/ANSI, weighted) 0.04% RMS (JIS/NAB, weighted)
Fast Wind Time:	85 seconds for C-60 tape
Motors:	1 FG Servo-controlled DC Capstan motor, 1 DC Reel motor and 1 DC Control motor
Recording Time:	15 minutes for C-60 tape
Dimension:	455(W) × 120(H) × 370(D) mm (17-15/16" × 4-3/4" × 14-9/16")
Weight:	
Net:	9 kg (20 lbs.)
Shipping:	10 kg (22 lbs.)

ELECTRICAL

MIXER SECTION

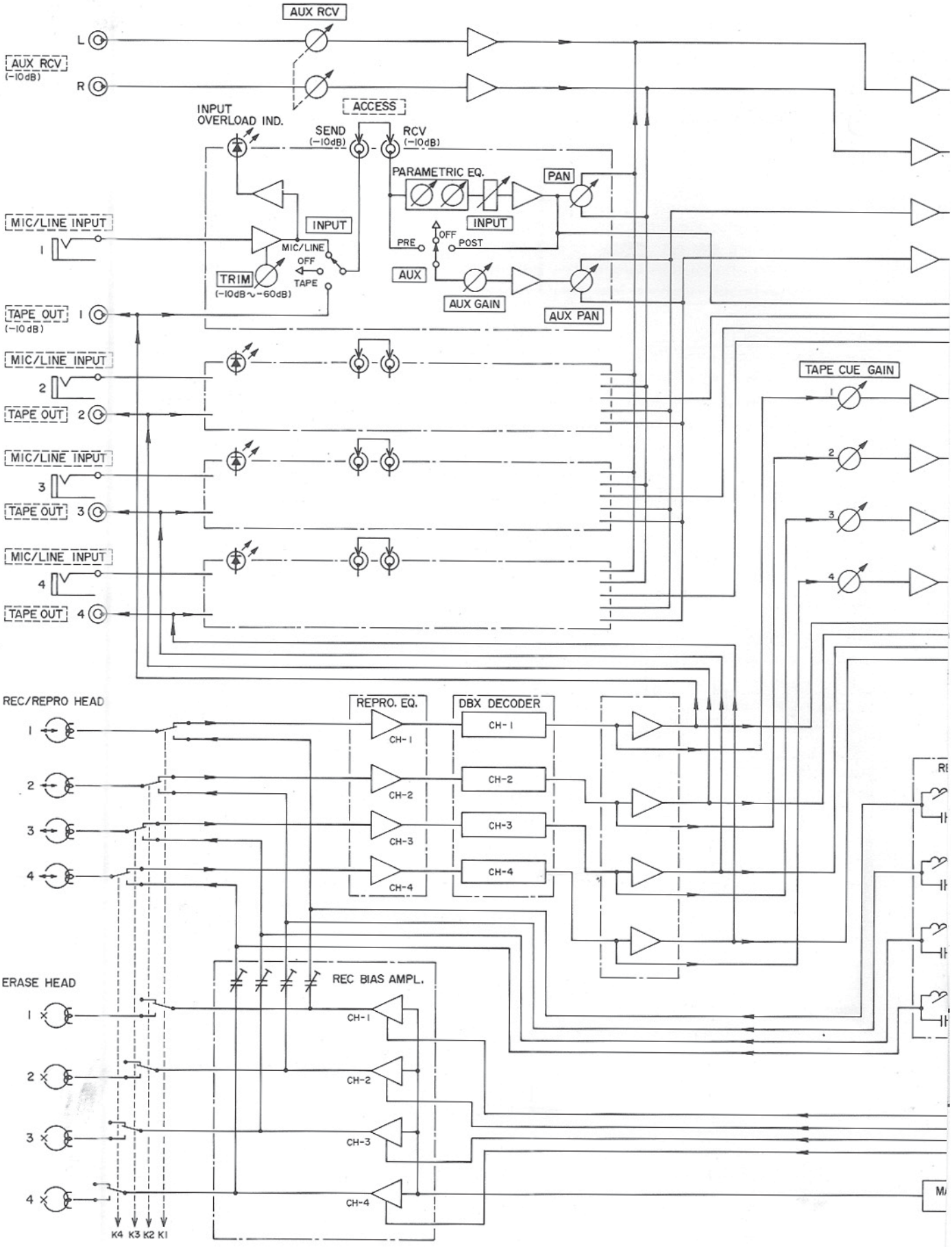
Mic/Line Input (X 4):	
Mic Impedance:	10k ohms or Less
Input Impedance:	60k ohms
Nominal Input Level:	Mic −60 dB (1 mV), at TRIM max. Line −10 dB (0.3 V), at TRIM min.
Minimum Input Level:	−68 dB (0.4 mV)
Maximum Input Level:	+15 dB (5.6 V)
Aux Rcv (X 2):	
Input Impedance:	37k ohms
Nominal Input Level:	−10 dB (0.3 V)
Access Send:	
Output Impedance:	100 ohms
Nominal Load Impedance:	10k ohms
Minimum Load Impedance:	1.7k ohms
Nominal Output Level:	−10 dB (0.3 V)
Maximum Output Level:	+15 dB (5.6 V)
Access Rcv:	
Input Impedance:	68k ohms
Nominal Input Level:	−10 dB (0.3 V)
Maximum Input Level:	+15 dB (5.6 V)
Line Out (X 2), Aux Out (X 2):	
Output Impedance:	100 ohms
Nominal Load Impedance:	10k ohms
Minimum Load Impedance:	1.7k ohms
Nominal Output Level:	−10 dB (0.3 V)
Maximum Output Level:	+15 dB (5.6 V)
Aux Send (X 2):	
Output Impedance:	100 ohms
Nominal Load Impedance:	10k ohms
Minimum Load Impedance:	1.7k ohms
Nominal Output Level:	−10 dB (0.3 V)
Maximum Output Level:	+15 dB (5.6 V)
Headphone Output (Stereo):	
Nominal Load Impedance:	8 ohms
Minimum Load Impedance:	4 ohms
Maximum Output Level:	100 mW at 8 ohms
Equalizer:	
Type:	Peak/Dip Parametric
Frequency:	Low/Mid, 62 Hz to 1.5k Hz Mid/High, 1k Hz to 8k Hz
Peak/Dip Level:	±15 dB
Tape Cue (Stereo):	
Output Impedance:	100 ohms
Nominal Load Impedance:	10k ohms
Minimum Load Impedance:	1.7k ohms
Nominal Output Level:	−10 dB (0.3 V)
Maximum Output Level:	+15 dB (5.6 V)
Input Overload Indicator:	
Indication Level:	22 dB above Nominal Input Level
Buss Overload Indicator:	
Indication Level:	10 dB above Nominal Output Level (10 dB above Nominal Recording Level)
Frequency Response:	20 Hz ~ 20,000 Hz ±1 dB
Signal-to-Noise Ratio:	
One mic in to line out:	68 dB (IHF, A weight) 65 dB (unweighted)
One line in to line out:	80 dB (IHF, A weight) 76 dB (unweighted)
Total Harmonic Distortion:	0.05 % at 1k Hz, nominal level
Crosstalk:	65 dB at 1k Hz

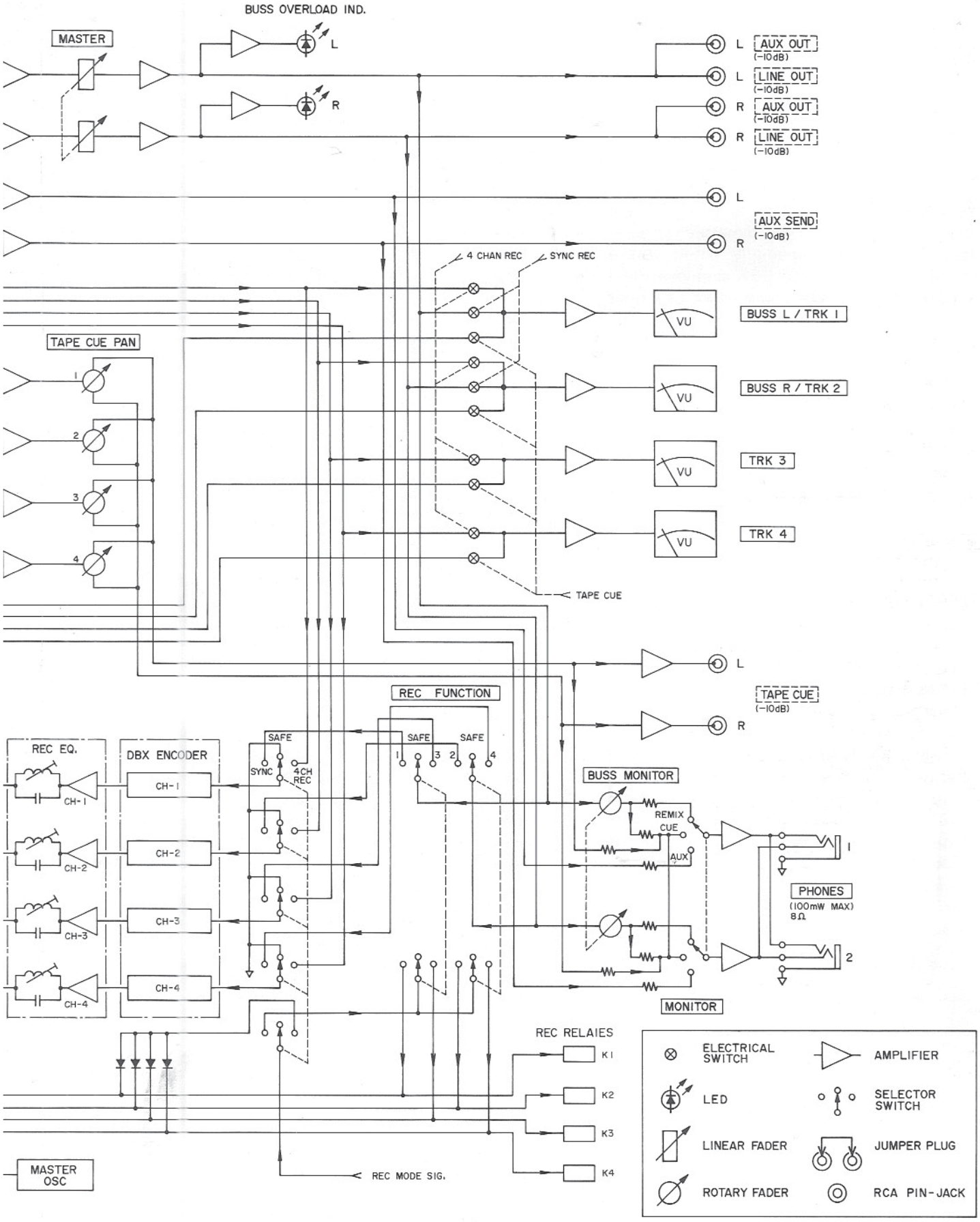
RECORDER SECTION

Record Channels:	4 with DBX Type-II NR full time Encoding
Playback Channels:	4 with DBX Type-II NR full time Decoding
Tape Out (4 Channel):	
Output Impedance:	100 ohms
Nominal Load Impedance:	10k ohms
Minimum Load Impedance:	1.7k ohms
Nominal Output Level:	−10 dB (0.3 V)
Maximum Output Level:	+15 dB (5.6 V)
Frequency Response:	40 Hz~14,000 Hz, ±3 dB at O VU (Rec/Repro)
Signal-to-Noise Ratio:	Referenced to 3% THD Level (285 nWb/m) at 315 Hz,
One line in to line out:	90 dB (IHF, A weighted) 75 dB (unweighted)
Total Harmonic Distortion:	1.5% at 315 Hz, OVU level (Rec/Repro)
Crosstalk:	70 dB at 1k Hz, O VU level
Erasure:	70 dB, referenced to 285 nWb/m at 1kHz
Power Requirment:	100/120/220/240V AC, 50/60Hz, 30W (General Export Model) 120V AC, 60Hz, 30W (U.S.A./Canada Model) 220V AC, 50Hz, 30W (Europe Model) 240V AC, 50Hz, 30W (U.K./Australia Model)
Optional Accessories:	
Model RC-30P:	Remote Punch-In/Out Foot Pedal.
Model 109B:	Designed to low impedance mics to high impedance inputs to virtually eliminate RF signals induced in the mic cable. The 109B is particulary useful when the mic cable length exceeds 15 feet.

Specifications subject to change without notice or obligation.
OdB=1 V.

BLOCK DIAGRAM




BUSS OVERLOAD IND.
MASTER
L
R
L AUX OUT (-10dB)
L LINE OUT (-10dB)
R AUX OUT (-10dB)
R LINE OUT (-10dB)
L
AUX SEND (-10dB)
R
4 CHAN REC
SYNC REC
VU
BUSS L / TRK 1
BUSS R / TRK 2
TRK 3
TRK 4
TAPE CUE PAN
1
2
3
4
TAPE CUE
L
TAPE CUE (-10dB)
R
REC FUNCTION
SAFE
SAFE
SAFE
1 3 2 4
REC EQ.
DBX ENCODER
SYNC
4CH REC
CH-1
CH-2
CH-3
CH-4
BUSS MONITOR
REMIX
CUE
AUX
PHONES (100mW MAX) 8Ω
MONITOR
REC RELAIES
K1
K2
K3
K4
REC MODE SIG.
MASTER OSC
ELECTRICAL SWITCH
AMPLIFIER
LED
SELECTOR SWITCH
LINEAR FADER
JUMPER PLUG
ROTARY FADER
RCA PIN-JACK

The DBX is a wide-band compression-expansion system which provides a net noise reduction (broadband, not just hiss) of a little more than 30 dB. In addition, the compression during recording permits a net gain in tape headroom of about 10 dB.

A compression factor of 2:1 is used before recording; then, 1:2 expansion on reproduce. These compression and expansion factors are linear in decibels and allow the system to produce tape recordings with over a 100 dB dynamic range — an important feature, especially when you're making live recordings. The DBX employs RMS level sensors to eliminate compressor-expander tracking errors due to phase shifts in the tape recorder, and provides excellent transient tracking capabilities.

To achieve a large reduction in audible tape hiss, without danger of overload or high frequency self-erasure on the tape, frequency pre-emphasis and de-emphasis are added to the signal and RMS level sensors.

If you're an electronics engineer, all of the above gab may tell you the whole story of what's going on in the DBX, but if you're not, to make things a little easier to understand we'll ask you to use your imagination.

Imagine four little recording engineers in the box with each of their hands on a volume control. They are incredibly fast but very stupid, so you must give them a set of rules. You tell them to raise signals that are below "O VU", and reduce signals that are higher than "O VU".

The lower the signal is, the more they raise it, and the higher levels above "O VU" get lowered more and more as they go up in level past "O". This is the 2:1 compression . You also tell them to call "0.316 V" "O VU". Here they do nothing, no charge except frequency pre-emphasis or boost. Since you know they are going to keep the high levels under control, you can raise the "top end" a bit and still not overload the tape. Just to keep it simple for them, the boost in highs is fixed. They put it in all the time no matter what level changes they are making. Now we play the tape back, and say OK, do everything backwards. Levels above "0.316 V" "O VU" are raised and levels below "0.316 V" are lowered, and while you're at it, fellows, take off the extra top end as well. Follow the rules in reverse. As long as you don't confuse them by shifting the "O VU" point, they work just great, but — don't put in more than "0.316 V" as zero VU, and don't make the tape playback zero anything other than "0.316 V" either. As we said they're very dumb and will follow instructions very precisely. Differing levels will produce decoding errors.

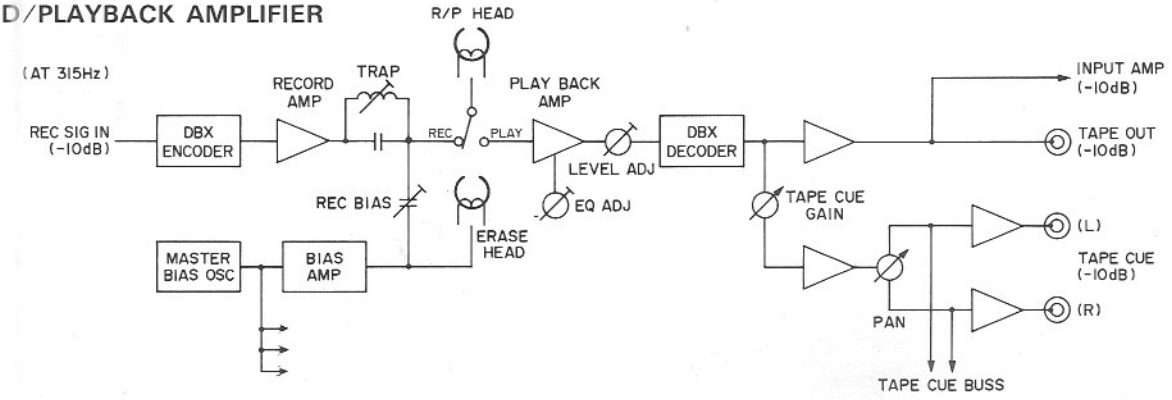
The reason these errors may not be objectionable is that people could have played or sung or whatever with a little more or less dynamics. A small change won't be as noticeable as a mistake, but it is not perfect. The tolerance here is not electronic, it's human. To get exactly what you put in, it is necessary to get an exact "O VU", 0.316 V in and out. The system is level sensitive although it is realistic to say it is "artistically" foregiving.

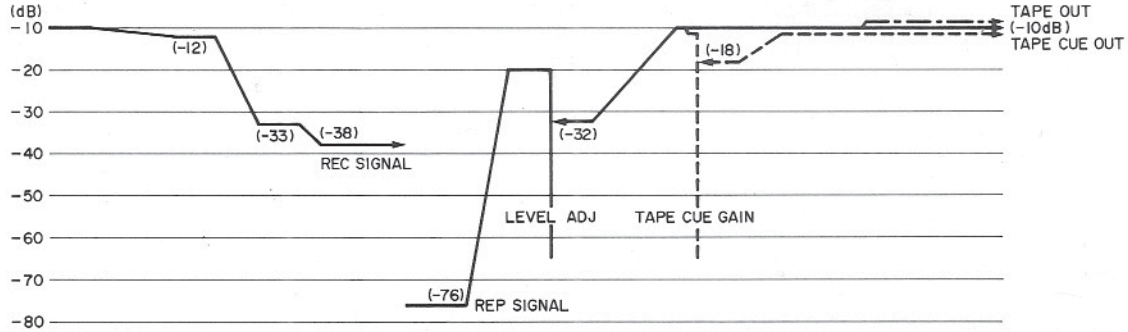
Subsonics and Interference

The DBX incorporates an effective bandpass filter with −3 dB response at 20 Hz and 30 kHz. This filter suppresses undesirable sub- and supersonic frequencies to keep them from introducing errors into the encode or decode process. However, if rumble from trains or trucks, is picked up by your microphone and fed to the DBX — filters are not perfect — modulation of the program material during low level passages may occur. This low frequency component will not itself be passed through the recorder and so, will not be present at reproduce for proper decoding. If this low level decoding error is encountered, and subsonics are suspected, we suggest the addition of a suitable high pass filter for 244 only — in the Microphone Line or at the ACCESS-SEND-RCV jacks ahead of the DBX and after the mic preamplifier for further attenuation of these subsonic frequencies.

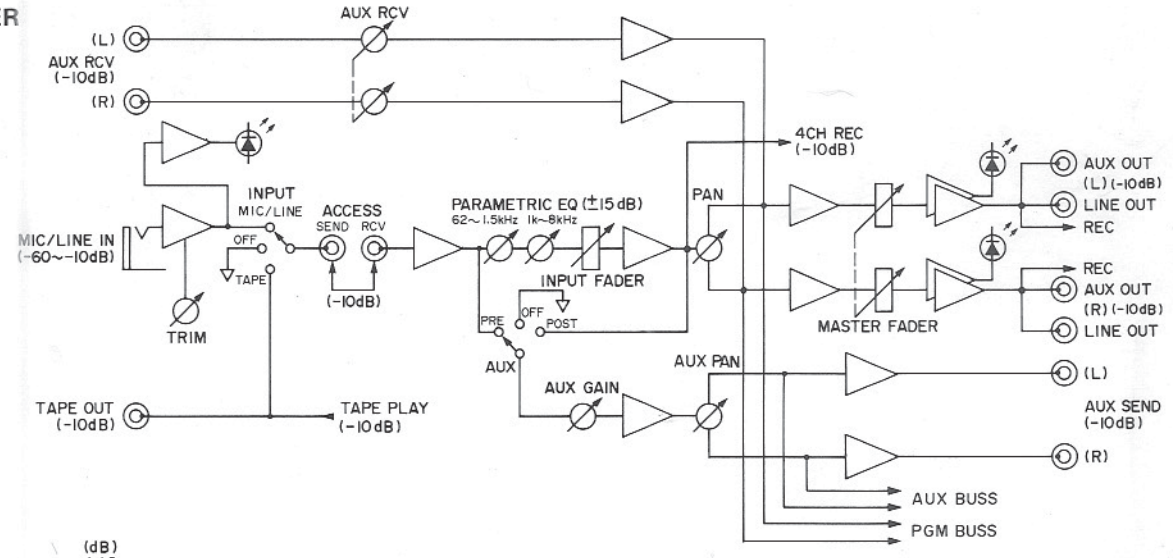
LEVEL DIAGRAM

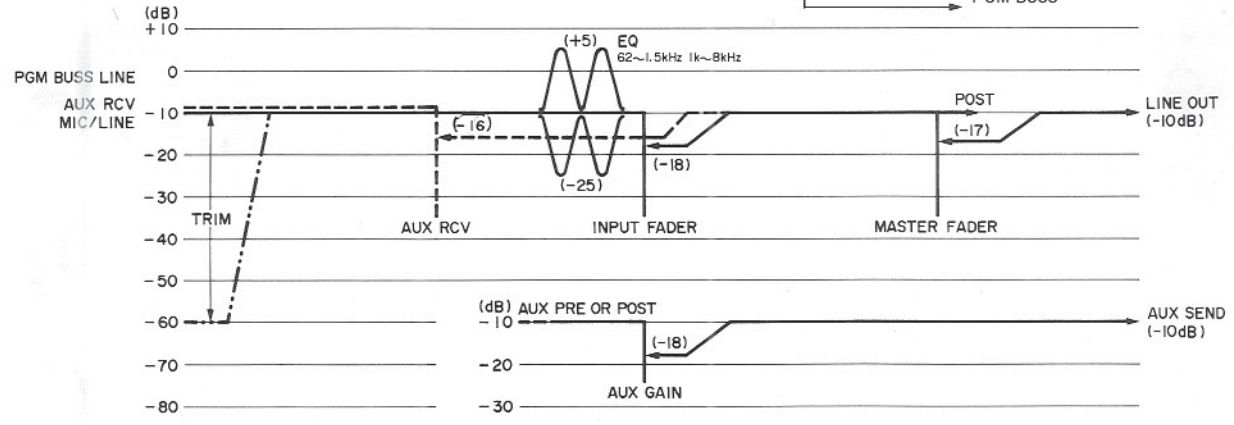
RECORD/PLAYBACK AMPLIFIER





MIXER





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244

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